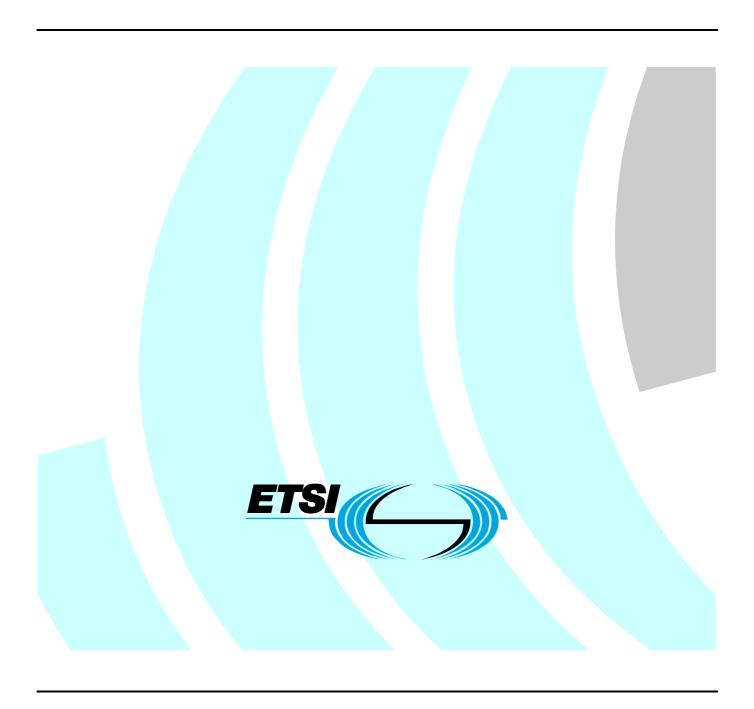
ETSITS 186 009-2 V2.1.1 (2009-03)

Technical Specification

Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN);
SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks;
Part 2: Test Suite Structure and Test Purposes (TSS&TP)



Reference DTS/TISPAN-06025-2-NGN-R2

Keywords

BICC, CTS, interworking, SIP, testing, TSS&TP

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

The present document is part 2 of a multi-part deliverable covering the Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks;

- Part 1: "Protocol Implementation Conformance Statement (PICS)";
- Part 2: "Test Suite Structure and Test Purposes (TSS&TP)";
- Part 3: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) proforma specification".

1 Scope

The present document specifies the Test Suite Structure and Test Purposes (TSS&TP) for SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN)subsystem and Circuit Switched (CS) networks ES 283 027 [1]. The references [1] and [16] are indentical.

A further part of the present document specifies the Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) proforma based on the present document.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

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NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[1]	ETSI ES 283 027 (V2.5.1): "Telecommunications and Internet converged Services and Protocols
	for Advanced Networking (TISPAN) Endorsement of the SIP-ISUP Interworking between the IP
	Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks
	[3GPP TS 29.163 (Release 7), modified]".

- [2] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+) Universal Mobile Telecommunications System (UMTS) Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 7.9.0 Release 7)".
- [3] ITU-T Recommendations Q.761 to Q.764 (2000): "Signalling System No.7 ISDN User Part (ISUP)".
- [4] Void.
- [5] ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".
- [6] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".
- [7] IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".
- [8] ISO/IEC 9646-1 (1994): "Conformance testing methodology and framework Part 1: General Concepts".

- [9] ISO/IEC 9646-3 (1992): "Conformance testing methodology and framework Part 3: The Tree and Tabular Combined Notation".
 [10] ISO/IEC 9646-7 (1994): "Conformance testing methodology and framework Part 7: Implementation Conformance Statement".
 [11] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".
 [12] Void.
- [13] ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".
- [14] ETSI TS 183 008: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR); Protocol specification".
- [15] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 8.5.0 Release 8)".
- [16] ETSI TS 129 527 (V8.2.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); TISPAN; Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks [3GPP TS 29.163 (Release 7), modified] (3GPP TS 29.527 version 8.2.0 Release 8)".
- [17] IETF RFC 3556: "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth".
- [18] IETF RFC 3264: "An Offer/Answer Model with Session Description Protocol (SDP)".
- [19] IETF RFC 4040: "RTP Payload Format for a 64 kbit/s Transparent Call".
- [20] ITU-T Recommendation F.182: "Operational provisions for the international public facsimile service between subscribers with Group 3 facsimile terminals (Telefax 3)".
- [21] ITU-T Recommendation F.184: "Operational provisions for the international public facsimile service between subscriber stations with group 4 facsimile terminals (telefax 4)".
- [22] ITU-T Recommendation F.230: "Service requirements unique to the mixed mode (MM) used within the teletex service".
- [23] ITU-T Recommendation F.220: "Service requirements unique to the processable mode number eleven (PM11) used within the teletex service".
- [24] ITU-T Recommendation F.200: "Teletex service".
- [25] ITU-T Recommendation F.300: "Videotex service".
- [26] ITU-T Recommendation F.60: "Operational provisions for the international telex service".
- [27] ITU-T Recommendation F.721: "Videotelephony teleservice for ISDN".
- [28] ETSI ETS 300 356-1: "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 1: Basic services [ITU-T Recommendations Q.761 to Q.764 (1999) modified]".
- [29] ITU-T Recommendation X.213: "Information technology Open Systems Interconnection Network service definition".
- [30] ISO/IEC 8348: "Information technology Open Systems Interconnection Network service definition".

[31]	$ITU-T\ Recommendation\ T.38:\ "Procedures\ for\ real-time\ Group\ 3\ facsimile\ communication\ over\ IP\ networks".$
[32]	ITU-T Recommendation Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".
[33]	ITU-T Recommendation Q.737.1: "Stage 3 description for additional information transfer supplementary services using Signalling System No. 7: User-to-user signalling (UUS)".
[34]	ITU-T Recommendation Q.734.1: "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Conference calling".
[35]	ITU-T Recommendation Q.734.2: "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Three-party service".

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area. For non-specific references, the latest version of the referenced document (including any amendments) applies.

Not applicable.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in SIP / ISUP interworking reference specification, in ISDN layer 3 reference specification, in ISO/IEC 9646-1 [8], in ISO/IEC 9646-3 [9], in ISO/IEC 9646-7 [10] and the following apply:

Abstract Test Case (ATC): complete and independent specification of the actions required to achieve a specific test purpose, defined at the level of abstraction of a particular Abstract Test Method, starting in a stable testing state and ending in a stable testing state

Abstract Test Method (ATM): description of how an SUT is to be tested, given at an appropriate level of abstraction to make the description independent of any particular realization of a Means of Testing, but with enough detail to enable abstract test cases to be specified for this method

Abstract Test Suite (ATS): test suite composed of abstract test cases

Implementation Under Test (IUT): implementation of one or more OSI protocols in an adjacent user/provider relationship, being part of a real open system which is to be studied by testing

Means of Testing (MOT): combination of equipment and procedures that can perform the derivation, selection, parameterization and execution of test cases, in conformance with a reference standardized ATS, and can produce a conformance log

PICS proforma: document, in the form of a questionnaire, which when completed for an implementation or system becomes the PICS

PIXIT proforma: document, in the form of a questionnaire, which when completed for the SUT becomes the PIXIT

Point of Control and Observation (PCO): point within a testing environment where the occurrence of test events is to be controlled and observed, as defined in an Abstract Test Method

pre-test condition: setting or state in the SUT which cannot be achieved by providing stimulus from the test environment

Protocol Implementation Conformance Statement (PICS): statement made by the supplier of a protocol claimed to conform to a given specification, stating which capabilities have been implemented

Protocol Implementation eXtra Information for Testing (PIXIT): statement made by a supplier or implementor of an SUT (protocol) which contains or references all of the information related to the SUT and its testing environment, which will enable the test laboratory to run an appropriate test suite against the SUT

SIP number: number conforming to the numbering and structure specified in ITU-T Recommendation E.164 [11]

System Under Test (SUT): real open system in which the SUT resides

user: access protocol entity at the User side of the user-network interface where a T reference point or coincident S and T reference point applies

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ATC Abstract Test Case ATM Abstract Test Method ATP Access Transport Parameter **ATS** Abstract Test Suite BCI **Backward Call Indicators CPS** Calling Party's Category Digital Subscriber System No. 1 DSS1 Forward Call Indicators FCI HLC High Layer Compatibility **ISDN** Integrated Services Digital Network **ISUP** ISDN User Part Implementation Under Test **IUT** Means Of Testing MOT

NCI Nature of Connection Indicators
OBCI Optional Backward Call Indicators
OFCI Optional Forward Call Indicator

PICS Protocol Implementation Conformance Statement
PIXIT Protocol Implementation eXtra Information for Testing

SUT System Under Test

TMR Transmission Medium Requirement

TP Test Purpose
TSS Test Suite Structure

TTCN Tree and Tabular Combined Notation

NOTE: The ISUP message acronyms can be found in table 2/ ITU-T Recommendation Q.762 [3].

4 Implementation under test and test methods

4.1 Identification of the system and implementation under test

FFS

5 Test Suite Structure (TSS)

The Test Suite Structure is in close alignment with ES 283 027 [1].

5.1 Interworking from SIP to ISUP (outgoing call)

SIP -ISUP Basic call		
	Sending of the Initial address message (IAM)	101xxx
	Sending of the Subsequent address message (SAM)	102xxx
	Sending of COT	103xxx
	Receipt of the Address complete message (ACM)	104xxx
	Receipt of the Call progress message (CPG)	105xxx
	Receipt of the answer message (ANM)	106xxx
	Receipt of the Connect message (CON)	107xxx
	Receipt of the Release message (REL)	108xxx
	Autonomous release at I-MGCF	109xxx
	Receipt of the BYE, CANCEL message / sending of a REL	110xxx
	message	
	Receipt of Reset circuit message (RSC), Circuit group reset	111xxx
	message (GRS) or Circuit group blocking message (CGB) with	
	the indication hardware failure oriented	
	Receipt of the SUSPEND Message (SUS)	111xxx
	Receipt of the RESUME Message (RES)	112xxx

Figure 1: Basic call Test suite structure for interworking between SIP to ISUP (outgoing call)

5.2 Interworking from ISUP to SIP (incoming call)

ISUP-SIP Basic call		
	Sending of the INVITE message	301xxx
	Receipt of the Subsequent address message (SAM)	302xxx
	Sending of the Address complete message (ACM)	303xxx
	Sending of the Call progress message (CPG)	304xxx
	Sending of the answer message (ANM)	305xxx
	Sending of the Connect message (CON)	306xxx
	Receipt of the Release message (REL)	307xxx
	Sending of the Release Message (REL)	308xxx
	Autonomous release	309xxx
	Receipt of Reset circuit message (RSC)	310xxx
	Receipt of Circuit group reset message (GRS)	311xxx
	Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	312xxx

Figure 2: Basic call Test suite structure for interworking between ISUP to SIP (incoming call)

5.3 Supplementary Services - Interworking from SIP to ISUP (outgoing call)

SIP-ISUP Supplementary Services		
	Calling Line Identification (CLI)	501xxx
	Call Hold (HOLD)	502xxx
	Terminal Portability (TP)	503xxx
	Conference Calling (CONF)	504xxx
	Three-Party (3PTY)	505xxx
	Connected Line Identification (COL)	506xxx
	Malicious call identification (MCID)	507xxx
	Subaddressing (SUB)	508xxx
	Call Diversion (CDIV)	509xxx
	Call Waiting (CW)	510xxx
	User to User Signalling (UUS)	511xxx
	Explicit Call transfer (ECT)	512xxx
	Completion of Call to Busy Subscriber (CCBS)	513xxx
	Completion of Calls on No reply (CCNR)	514xxx
	Anonymous Call Rejection (ACR)	515xxx
	Closed user group (CUG)	516xxx

Figure 3: Supplementary Services Test suite structure for interworking between SIP to ISUP (outgoing call)

5.4 Supplementary Services - Interworking from ISUP to SIP (incoming call)

ISUP-SIP		
	Calling Line Identification (CLI)	601xxx
	Call Hold (HOLD)	602xxx
	Terminal Portability (TP)	603xxx
	Conference Calling (CONF)	604xxx
	Three-Party (3PTY)	605xxx
	Connected Line Identification (COL)	606xxx
	Subaddressing (SUB)	607xxx
	Closed User Group (CUG)	608xxx
	Call Diversion (CDIV)	609xxx
	User to User Signalling (UUS)	610xxx
	Explicit Call transfer (ECT)	611xxx
	Anonymous Call Rejection (ACR)	612xxx
	Call waiting (CW)	613xxx
	Malicious call identification (MCID)	614xxx

Figure 4: Supplementary Services Test suite structure for interworking between ISUP to SIP (outgoing call)

6 Test purposes (TP)

6.1 Introduction

For each test requirement a Test Purpose (TP) is defined.

6.1.1 Test purpose (TP) naming convention

For each test requirement a Test Purpose (TP) is defined.

All test purposes belong to the main group ISUP_SIP_Interworking. Groups are organized according to the test suite structure (TSS). Each test purpose is presented in a separate table. The first row of the table contains the following items:

TP Identifier of the test purpose;

SIP reference the reference to the requirement in the DSS1 layer 3 Recommendation, which led to the TP;

ISUP reference the reference to the requirement in the interworking specification and the requirement in the

SIP-UP Recommendation, which led to the TP.

6.1.2 Source of test purpose definition

The test purposes have been developed based on ES 283 027 [1] as an endorsement of TS 129 163 [15].

6.1.3 Test purpose structure

The test purpose structure is according to the test suite structure (TSS).

6.2 Test purposes for the basic call

6.2.1 Interworking from SIP to ISUP (Outgoing Call)

6.2.1.1 Sending of the Initial Address Message (IAM)

TP101001	SIP reference: RFC 3261	[6]	ISUP reference:	
			ES 283 027 [1], clause 7.2.3.1.	.1
TSS reference:	SIP-ISUP/Basic call/ Sending of the	e Initial Add	dress message (IAM)/	
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Normal call setup without precondi	tion reequii	irement	
			is not included in the Supported or Rec	
			diately after the reception of the INVITE	=, The
	I-MGCF shall set the continuity ind	cators to "	Continuity check not required".	
SIP Parameter				
values:				
ISUP Parameter				
values:				
Comments:	SIP	SL	UT ISUP	
	INVITE →		→ IAM	
	180 Ringing ←		← ACM	
		Ringing	ng tone	
	200 OK INVITE ←		← ANM	
	ACK →			
		Conver	ersation	
	BYE →		→ REL	
	200 OK BYE ←		← RLC	

TP101002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.1	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Add		
SIP selection	PICS 4/4 AND PICS 4/5		
criteria:			
ISUP selection			
criteria:		d been dear and a mean difference of the left	
Test purpose:	Call setup with precontion tag in the Supported header and preconditions are fullfield successful Ensure if a Continuity Check procedure is supported in the ISUP network and SIP precondition extension are included in the SIP Supported header and the preconditions are indicated as fullfield in the SDP, the I-MGCF shall send the IAM immediately after the		
SIP Parameter	reception of the INVITE. The preconditions me INVITE: Supported: 100rel, precondition	et is sent in the 200 OK INVITE.	
values:	SDP a=curr:qos local sendrecv		
values.	a=curr:qos remote none		
	a=des:qos mandatory local sen	drecv	
	a=des:qos none remote sendrecv		
	200 OK INVITE		
	SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv		
	a=des:gos mandatory local sen	drecv	
	a=des:qos mandatory remote s		
ISUP Parameter	IAM: Continuity indicator: Continuity check no	required	
values:			
Comments:	SIP SU		
	INVITE -	→ IAM	
	180 Ringing ←	← ACM	
	Ringin	•	
	200 OK INVITE ← ACK →	← ANM	
	Conve	reation	
	BYE -	→ REL	
	200 OK BYE	€ RLC	
	200 ON BIL	ILLO	

TP101003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.1	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Add	dress message (IAM)/	
SIP selection criteria:	NOT PICS 4/4 AND NOT PICS 4/5		
ISUP selection			
criteria:			
Test purpose:	Call setup with precontion tag in the Supported header and preconditions are fullfield unsuccessful Ensure if the received SDP indicates that precondition is fulfilled the I-MGCF shall set the continuity indicators to "continuity check is not required". The SUT does not an answer to		
	the precondition requirement.		
SIP Parameter	INVITE: Supported: 100rel, precondition		
values:	SDP a=curr:qos local sendrecv		
	a=curr:qos remote none		
	a=des:qos mandatory local sendrecv		
	a=des:qos none remote sendrecv		
ISUP Parameter	IAM: Continuity indicator: Continuity check not required		
values:			
Comments:	SIP SI	JT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Ringin	g tone	
	200 OK INVITE	← ANM	
	ACK →		
	Conve	rsation	
	BYE →	→ RFL	
	200 OK BYE ←	← RLC	

TP101004	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.1	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/		
SIP selection criteria:	PICS 4/4 AND PICS 4/5		
ISUP selection criteria:	PICS 4/1		
Test purpose:	Call setup with precondition tag in the Require header and requirement for recource reservation Ensure if the INVITE request contains the precondition tag in the Require header the		
		is not fulfilled the I-MGCF shall set the continuity on a previous circuit" or "required on this circuit".	
SIP Parameter values:	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv		
	183: Require: 100rel SDP		
	UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv		
	200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv		
ISUP Parameter values:	IAM: "continuity check required on this ci previous circuit"	rcuit" or "Continuity check performed on a	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ←	SUT ISUP → IAM	
	UPDATE → 200 OK UPDATE ←	→ COT	
	180 Ringing ← PRACK → 200 OK PRACK ←	← ACM inging tone	
	200 OK INVITE ← ACK →	← ANM onversation	
	BYE → ← 200 OK BYE ←	→ REL ← RLC	

TP101005	SIP reference: F	RFC 3261 [6]		ISUP reference: 027 [1], clause 7.2.3.1.1	
TSS reference:	SIP-ISUP/Basic call/ Se	nding of the Initial A			
SIP selection	PICS 4/4 AND PICS 4/5		-	,	
criteria:					
ISUP selection	PICS 4/1				
criteria:	Call setup with precondition tag in the Supported header and requirement for recource				
Test purpose:	reservation Ensure if the INVITE red	quest contains the p	recondition tag in	the Supported header the	
				MGCF shall set the continuity	
SIP Parameter			a previous circu	it" or "required on this circuit".	
values:	SDP a=curr:qo: a=curr:qo: a=des:qos a=des:qos	a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv			
	SDP a=curr:qo: a=curr:qo: a=des:qo: a=des:qo:	a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv			
	a=curr:qo: a=des:qos				
	a=curr:qo: a=des:qos				
ISUP Parameter values:	IAM: "continuity check re previous circuit"	equired on this circu	it" or "Continuity	check performed on a	
Comments:	INVITE 183 Session Progress PRACK 200 OK PRACK	→ ← → ←	TUS	ISUP IAM	
	UPDATE 200 OK UPDATE	→	→	СОТ	
	180 Ringing PRACK 200 OK PRACK	← → ←	←	ACM	
	200 OK INVITE ACK	← →	ing tone ersation	ANM	
	BYE 200 OK BYE	→ ←	→ ←	REL RLC	

TP101006	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clause 7.2.3.1.1
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Add	dress message (IAM)/
SIP selection	NOT PICS 4/4 AND PICS 4/5	
criteria:		
ISUP selection	PICS 4/1	
criteria:		
Test purpose:	Call setup with precondition tag in the Require reservation	e header and requirement for recource
	Ensure if the INVITE request contains the pre- received SDP indicates that precondition is no provisional response if preconditions are not s	ot fulfilled the I-MGCF shall send a 5xx final
SIP Parameter	INVITE: Require: precondition	••
values:	SDP a=curr:qos local none	
	a=curr:qos remote none	
	a=des:qos mandatory local sen	
	a=des:qos none remote sendre	CV
ISUP Parameter values:		
Comments:	SIP SU	JT ISUP
	INVITE -	
	580 Precondition Failure ←	
	ACK →	

TP101007	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.1	
T00 (RFC 3264 [18]	
TSS reference: SIP selection	SIP-ISUP/Basic call/ Sending of the Initial A	ddress message (IAM)/	
criteria:	PICS 4/4 AND PICS 4/5		
ISUP selection	PICS 1/3 AND NOT PICS 4/1		
criteria:			
Test purpose:	COT procedure not supported, IAM delayed	l until preconditions met	
	Francis Continuity Charles are advers in not	aumanded in the ICLID nativisals, and the CDD	
	in the received INVITE request contains pre	supported in the ISUP network, and the SDP conditions not met, the I-MGCF shall delay	
	sending the IAM until the SIP preconditions	are met and set the continuity indicators in the	
	resulting IAM to "Continuity check not require	red".	
SIP Parameter	INVITE: Require: precondition		
values:	SDP a=curr:qos local none a=curr:qos remote none		
	a=des:qos mandatory local se	endrecy	
	a=des:qos none remote send		
	183: Require: 100rel SDP a=curr:qos local none		
	a=curr:qos remote none		
	a=des:qos mandatory local sendrecv		
	a=des:qos mandatory remote	sendrecv	
	a=conf:qos remote sendrecv		
	UPDATE:		
	SDP a=curr:qos local sendrecv		
	a=curr:qos remote none		
	a=des:qos mandatory local se		
	a=des:qos mandatory remote	Sendrecv	
	200 OK UPDATE		
	SDP a=curr:qos local sendrecv		
	a=curr:qos remote sendrecv	andragy.	
	a=des:qos mandatory local se a=des:qos mandatory remote		
ISUP Parameter	IAM Continuity Indicator: continuity check re		
values:	COT Continuity Indicator: continuity chec		
Comments:		SUT ISUP	
	INVITE ->		
	183 Session Progress ← PRACK →		
	200 OK PRACK		
	UPDATE →	→ IAM	
	200 OK UPDATE ←		
	180 Ringing ←	← ACM	
	PRACK 200 OK PRACK ←		
		ing tone	
	200 OK INVITE	← ANM	
	ACK →		
	Conv	versation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP101008	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.1 RFC 3264 [18]	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Ad	ddress message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5		
ISUP selection criteria:	PICS 1/3 AND NOT PICS 1/4 AND PICS 4/1		
Test purpose:	Media type not supported, call setup rejected Ensure that the I-MGCF shall reject an INVIT unsupported media types by sending a statu	TE request for a session only containing	
SIP Parameter values:	SDP: media type not supported in the SUT (PIXIT)		
ISUP Parameter values:			
Comments:	SIP INVITE → 488 Not Acceptable Here ← ACK →	BUT ISUP	

TP101009	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.3.3.1.1
		RFC 3264 [18]
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Ad	ddress message (IAM)/
SIP selection		
criteria:		
ISUP selection		
criteria:	2017	
Test purpose:	SUT rejects unsupported media types	
	Ensure that If several media streams are cor	stained in a single INVITE request, the
	I-MGCF shall select one of the supported me	1 ,
	media stream, and reject the other media str	
	answer, as detailed in RFC 3264 [18]. If sup	ported audio media stream(s) and supported
	non-audio media stream(s) are contained in	a single INVITE request, an audio stream
	should be selected.	
SIP Parameter	Offer: m=audio 4711 RTP/AVP 8	
values:	m= video 4713 RTP/AVP 31	
	Answer: m=audio 4711 RTP/AVP 8	
	m=video 0 RTP/AVP 31	
ISUP Parameter		
values:		
Comments:	SIP	SUT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
		ng tone
	200 OK INVITE ←	← ANM
	ACK →	
		ersation
	BYE →	→ REL
	200 OK BYE ←	← RLC

TP101010	SIP reference: RFC	3261 [6]		283 0	SUP reference: 27 [1], clause 7.2.3.1.1 RFC 3264 [18]
TSS reference:	SIP-ISUP/Basic call/ Sendi	ng of the	Initial Address me	ssage	(IAM)/
SIP selection	PICS 4/15				
criteria:					
ISUP selection					
criteria:					
Test purpose:	To tag included in 183 prov	visional re	sponse		
	Ensure that The I-MGCF st response, in order to estab				ckward non-100 provisional n RFC 3261 [6]
SIP Parameter	183 To taq included				
values:					
ISUP Parameter	ACM: oBCi "inband info ava	ailable"			
values:					
Comments:	SIP		SUT		ISUP
	INVITE	→		→	IAM
	183 Session Progress	(←	ACM(no indication)
	180 Ringing	←		←	CPG(Alerting)
			Ringing tone		
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	→		→	REL
	200 OK BYE	+		+	RLC

TP101011	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clause 7.2.3.1.1
		RFC 3264 [18]
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial	Address message (IAM)/
SIP selection		- · ·
criteria:		
ISUP selection		
criteria:		
Test purpose:	To tag included in 180 provisional respons	se
		o tag in the first backward non-100 provisional
	response, in order to establish an early dis	alog as described in RFC 3261 [6]
SIP Parameter	180 To tag included	
values:		
ISUP Parameter		
values:		
Comments:	SIP	SUT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Rii	nging tone
	200 OK INVITE ←	← ANM
	ACK →	
	Co	nversation
	BYE →	→ REL
	200 OK BYE ←	← RLC

TP101012	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clauses 7.2.3.1.2.2 and 7.2.3.1.2.3
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Add	dress message (IAM)/
SIP selection	PICS 2/1	
criteria:		
ISUP selection	PICS NOT 4/16	
criteria:		
Test purpose:	Setting of nature of connection indicator and t	orward call indicator
	Ensure that the SUT on receipt of an INVITE	message:
	sends an IAM message, where the Calling pa subscriber", the Nature of Connection Indica	ators (NCI) encoded as follows:
	Satellite indicator set to: One satellite circ	
	Echo control device indicator set to: "Out The Forward call indicator is encoded a	
	Interworking indicator: Interworking encoded a	
	ISUP/BICC Indicator: ISDN User part/BIC	
	ISUP/BICC Preference indicator: ISDN us	
	ISDN access indicator: Originating acces	
SIP Parameter		
values:		
ISUP Parameter	Nature of Connection Indicators (NCI):	
values:	Satellite indicator set to: "One satellite circ	
	Echo control device indicator set to: "Outg	oing echo control device included"
	Forward Call Indicators (FCI):	
	Interworking indicator: interworking encou	nterd
	ISDN user part indicator: ISDN user part/E	
	ISDN access indicator: originating access	
	ISDN user part preference indicator: ISDN	
Comments:	SIP St	
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringin	
	200 OK INVITE ←	← ANM
	ACK →	
	Conve	
	BYE -	→ REL
	200 OK BYE ←	← RLC

TP101013	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clause 7.2.3.1.2.2 Q.767
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Add	-4
SIP selection	PICS 1/1	arece meseage (ii iii),
criteria:		
ISUP selection		
criteria:		
Test purpose:	Setting of nature of connection indicator and t	forward call indicator T38 codec received
	E	''I ODD I' TOO
	Ensure that the SUT on receipt of an INVITE	message with SDP m line 1:38:
	sends an IAM message, where the Calling pa	
	subscriber", the Nature of Connection Indica	
	Satellite indicator set to: "One satellite False central devices indicator set to:	
	 Echo control device indicator set to: the Forward call indicator is encode 	"Outgoing echo control device not included".
	Interworking indicator: Interworking e	
	ISUP/BICC Indicator: ISDN User par	
		ON user part/BICC not required all the way
	ISDN access indicator: Originating a	
SIP Parameter	INVITE with SDP m line T:38	
values:		
ISUP Parameter	Nature of Connection Indicators (NCI):	
values:	Satellite indicator set to: "One satellite circ	cuit in the connection"
	Echo control device indicator set to: "Outg	oing echo control device not included"
	Forward Call Indicators (FCI):	
	Interworking indicator: interworking encou	
	ISDN user part indicator: ISDN user part/E	
	ISDN access indicator: originating access	
0	ISDN user part preference indicator: ISDN	
Comments:	SIP SU	
	INVITE →	→ IAM ← ACM
	100 1 11191119	
	Ringin 200 OK INVITE ←	g tone ← ANM
	ACK →	AINIVI
	Conve	reation
	BYE →	→ REL
	200 OK BYE	← RLC
	ZOU ON DIL	NLO NLO

TP101014	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.3 Q.767
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Ad	dress message	(IAM)/
SIP selection	PICS 2/3		
criteria:			
ISUP selection	PICS 4/16		
criteria:			1000
Test purpose:	Setting of nature of connection indicator and forward call indicator indicating ISDN and TMR 64 kBit/s Ensure that the SUT on receipt of an INVITE message with SDP m line CLEARMODE:		
	Linsure that the 301 off feceipt of all five 112	message with c	III III e CLLANWODE.
	 sends an IAM message, where the Calling subscriber", if the TMR = 64 kBit/s unresund Indicators (NCI) encoded as follows: the Nature of Connection Indicators 	stricted is used t	he Nature of Connection
	- Satellite indicator set to: "One satell		
	- Echo control device indicator set to:		
	 the Forward call indicator is encoded 	led as follows:	
	Interworking indicator: No interworki		
	ISUP/BICC Indicator: ISDN User pa		
	ISUP/BICC Preference indicator: IS ISDN access indicator: Originating a		CC not required all the way
SIP Parameter	TODIV access indicator. Originating a	icces iobiv.	
values:			
ISUP Parameter	Nature of Connection Indicators (NCI):		
values:	Satellite indicator set to: "One satellite cir-	cuit in the conne	ection"
	Echo control device indicator set to: o	utgoing echo co	ntrol device not included
	Forward Call Indicators (FCI):		
	Interworking indicator: No interworking en		
	ISDN user part indicator: ISDN user part/ ISDN access indicator: originating access		e way
	ISDN access indicator: originating access ISDN user part preference indicator: ISDN		not required all the way
Comments:		UT	ISUP
	INVITE →	→	IAM
	180 Ringing ←	←	ACM
		ng tone	
	200 OK INVITE ←	←	ANM
	ACK →		
	Conve	ersation	
	BYE →	→	REL
	200 OK BYE ←	+	RLC

TP101015	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.5	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Add	dress message (IAM)/	
SIP selection criteria:	Based on table 1	•	
ISUP selection criteria:			
Test purpose: SIP Parameter values: ISUP Parameter	Mapping of SDP into the TMR Ensure that the SUT in the Idle state on receipt of an INVITE message containing the media description defined in table 1 with the "a =" "b =" and "m=" lines set to a_b_m_LINE_VALUE: sends an IAM message, with the Transmission Medium Requirement (TMR) parameter set to TMR_VALUE. INVITE; a_b_m_LINE_VALUE IAM; TMR: ISUP_TMR		
values:	loin oi	IT IOUD	
Comments:	INVITE 180 Ringing ←	JT ISUP → IAM ← ACM g tone	
		← ANM rsation	
	BYE → 200 OK BYE ←	→ REL ← RLC	

Table 1

	1			es for test purposes TP1 LINE VALUE	01015	
		m= line	a_p_m	LINE_VALUE b= line	a= line	TMR_VALUE
test purposes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidt h-value=""></bandwidt></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt>	TMR codes
VA_01	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMU/8000</dynamic-pt>	"3,1KHz audio"
VA_02	audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	"3,1KHz audio"
VA_03	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>	"3,1KHz audio"
VA_04	audio	RTP/AVP	Dynamic PT	AS: 64 kbit/s	rtpmap: <dynamic-pt> CLEARMODE/8000 (NOTE 2)</dynamic-pt>	"64 kbit/s unrestricted"
VA_05	image	udptl	t38 [31]	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.3 8 [31]	"3,1 kHz audio"
VA_06	image	tcptl	t38 [31]	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.3 8 [31]	"3,1 kHz audio"
VA_07	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	"3,.1KHz audio"

NOTE 1: In this table the codec G.711 is used only as an example. Other codecs are possible.

TP101016	SIP re	eference: RFC 3261 [6]	_	SUP reference: 27 [1], clause 7.2.3.1.2.5		
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/						
SIP selection criteria:	PICS 2/3						
ISUP selection criteria:	PICS 4/18 AN						
SIP Parameter values:	Fallback connection type supported: Mapping of the second PSTN XML BearerCapability elements into TMR and USI prime Ensure that when the INVITE request includes multiple PSTN XML bearer information element: If the first stated codec in the INVITE is a codec appearing in table 1 and is the equivalent as stated within the second Bearer Capability in the XML Bearer Capability element then the I-MGCF shall map the XML Bearer Capability element into the TMR and USI prime and shall set the TMR to "64 kBit/s preferred". first BC: 3,1 kHz audio or speech second BC: unrestricted digital information with tones/announcements						
values:	USI Prime: TMR:	unrestricted digital info 64 kBit/s preferred	, i i i a i a i a i				
Comments:	SIP		SU	T	ISUP		
	INVITE	→		→	IAM		
	180 Ringing	←		←	ACM		
			Ringing				
	200 OK INVIT	INVITE ← ← ANM					
	ACK	→	_				
		_	Convers				
	BYE	→		→	REL		
	200 OK BYE	<u> </u>			RLC		

TP101017	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.5a					
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/						
SIP selection	PICS 4/18						
criteria:	DIOC 4/40						
ISUP selection criteria:	PICS 4/19						
Test purpose:	Fallback connection type supported: Mapping elements into USI and TMR prime	g of the first PSTN XML BearerCapability					
	Ensure when the INVITE request includes mu I-MGCF shall:	Iltiple PSTN XML BearerCapability then the					
	If the second stated codec in the INVITE is a						
	equivalent as stated within the first Bearer Ca element then the I-MGCF shall map the XML						
	prime and USI and shall map the TMR prime						
	(InformationTransferCabability).						
SIP Parameter	first BC: 3,1 kHz audio or speech						
values:	second BC: unrestricted digital information w	ith tones/announcements					
ISUP Parameter	USI: 3,1 kHz audio or speech						
values:	TMR prime: 3,1 kHz audio or speech						
Comments:	SIP SI	JT ISUP					
	INVITE →	→ IAM					
	180 Ringing ←	← ACM					
	Ringin	g tone					
	200 OK INVITE ←	← ANM					
	ACK →						
	Conve	rsation					
	BYE →	→ REL					
	200 OK BYE ←	← RLC					

TP101018	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.5a				
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/					
SIP selection criteria:	PICS 4/18					
ISUP selection criteria:	PICS 4/19					
Test purpose:	Fallback supported. Discard the second PSTN XML BearerCapability element if it is not equivalent to the first codec in the SDP					
	Ensure when the INVITE request includes mu MGCF shall:	ultiple PSTN XML BearerCapability then the I-				
	if the compared first codec stated within the INVITE is not equivalent as stated within the second XML Bearer Capability element, then the second XML Bearer Capability element shall be discarded.					
SIP Parameter	PSTN XML BC 1 (speech or 3,1 kHz audio)					
values:	PSTN XML BC 2 (unrestricted digital information	tion with tones and announcements)				
	SDP: m =audio xxx, RTP/AVP 0 8					
ISUP Parameter	USI: not included					
values:	TMR. 3,1 kHz audio					
Comments:	SIP SI	JT ISUP				
	INVITE →	→ IAM				
	180 Ringing ←	← ACM				
	Ringin	g tone				
	200 OK INVITE ←	← ANM				
	ACK →					
	Conve					
	BYE →	→ REL				
	200 OK BYE ←	← RLC				

TP101019	SIP reference: RF	C 3261 [6]		SUP reference:			
			ES 283 0	27 [1], clause 7.2.3.1.2.5			
TSS reference:	SIP-ISUP/Basic call/ Sendi	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/					
SIP selection	PICS 4/18						
criteria:							
ISUP selection							
criteria:							
Test purpose:	Mapping of PSTN XML Be	arerCapability elen	nent into the US	SI parameter			
	Ensure that the SUT in the	Idle state on recei	pt of an INVITE	message containing the XML			
	BearerCapability element t	he mapping of the	USI ISUP_USI	shall be taken from the PSTN			
	XML BearerCapability valu	e ISDN_BC.					
SIP Parameter	PSTN XML BearerCapabili	ty ISDN_BC					
values:	·						
ISUP Parameter	IAM: USI = ISUP_USI						
values:							
Comments:	SIP	SI	JT	ISUP			
	INVITE	→	→	IAM			
	180 Ringing	←	←	ACM			
		Ringin	g tone				
	200 OK INVITE	←	←	ANM			
	ACK	→					
		Conve	rsation				
	BYE	→	→	REL			
	200 OK BYE	-	+	RLC			

Values and selection criteria for the test purpose TP101019					
VA_01	ISDN_BC = speech	ISUP_USI = speech			
VA_02	ISDN_BC = 3,1 kHz audio	ISUP_USI = 3,1 kHz audio			
VA 03	ISDN BC = Unrestricted digital information	ISUP USI = Unrestricted digital information			

TP101020	SIP reference: RFC 3261 [6]	ES 2	-	SUP reference: 27 [1], clause 7.2.3.1.2.5		
TSS reference:	SIP-ISUP/Basic call/ Sending of the Ini	tial Address mes	sage	(IAM)/		
SIP selection	PICS 4/18					
criteria:						
ISUP selection						
criteria:						
Test purpose:	Mapping of PSTN XML BearerCapabil	ity element into th	ne TM	R parameter		
	Ensure that the SUT in the Idle state on receipt of an INVITE message containing one PSTN XML BearerCapability value ISDN_BC_ITC sends an IAM message, with the Transmission Medium Requirement (TMR) parameter set to ISUP_TMR.					
SIP Parameter	INVITE; PSTN XML BearerCapability					
values:						
ISUP Parameter values:	IAM: TMR					
Comments:	SIP	SUT		ISUP		
	INVITE →		→	IAM		
	180 Ringing ←		←	ACM		
		Ringing tone				
	200 OK INVITE ←		←	ANM		
	ACK →					
		Conversation				
	BYE →		→	REL		
	200 OK BYE ←		←	RLC		

Values and selection criteria for the test purpose TP101020					
VA_01	ISDN_BC_ITC = speech	ISUP_TMR = speech			
	ISDN_BC_ITR = 64 kbits/s				
VA_02	ISDN_BC_ITC = 3,1 kHz audio	ISUP_TMR = 3,1 kHz audio			
	ISDN_BC_ITR = 64 kbits/s				
VA_03	ISDN_BC_ITC = unrestricted digital information	ISUP_TMR = 64 kbits/s unrestricted			
	ISDN_BC_ITR = 64 kbits/s				

TP101021	SIP reference: RFC 3261 [6]	ISUP reference:					
11 101021		ES 283 027 [1], clause 7.2.3.1.2.5					
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/						
SIP selection	PICS 4/18	, , , , , , , , , , , , , , , , , , ,					
criteria:							
ISUP selection							
criteria:							
Test purpose:	No PSTN XML received, mapping of HLC in	ATP					
	 description defined in table 2 with the "a =" "b =" and "m=" lines to lines set to a_b_m_LINE_VALUE: sends an IAM message with the Access transport parameter containing the HLC information element. 						
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE						
ISUP Parameter	IAM; Access transport parameter HLC: H	LC_VALUE					
values:							
Comments:	SIP	SUT ISUP					
	INVITE →	→ IAM					
	180 Ringing ←	← ACM					
	_	ing tone					
	200 OK INVITE ←	← ANM					
	ACK →						
		rersation					
	BYE →	→ REL					
	200 OK BYE ←	← RLC					

Table 2

			Valu	es for test purposes TP10102	21	
M= line				b= line	a= line	HLC parameter HLC VALUE
Test purposes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidth- value></bandwidth- </modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt>	HLC_VALUE
				see note 1		
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	See note 2
VA_02	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMU/8000</dynamic-pt>	See note 2
VA_03	Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	See note 2
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>	See note 2
VA_05	Image	Udptl	t38	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.38 [31]	"Facsímile Group 2/3"
VA_06	Image	Tcptl	t38	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.38 [31]	"Facsímile Group 2/3"

NOTE 1: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.

NOTE 2: HLC normally absent in this case. It is possible for HLC to be present with the value "Telephony", although clause 6.3.1/ITU-T Recommendation Q.939 [13] indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

TP101022	SIP reference: RFC 3261 [6]		IP reference: [1], clause 7.2.3.1.2.10				
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/						
SIP selection criteria:		J (,				
ISUP selection criteria:							
Test purpose:	Mapping of PSTN XML BearerCapability	nd HighLayerCompat	tibility in ATP HLC				
	Ensure that the SUT in the Idle state on receipt of a INVITE message with a PSTN XML BearerCapability BC_VALUE and the PSTN XML HighLayerCompatibility HLC_VALUE sends an IAM message with the Access transport parameter containing the received PSTN XML HighLayerCompatibility].						
SIP Parameter	INVITE						
values:	PSTN XML BearerCapability: BC_VALUE PSTN XML HighLayerCompatibility: HLC	_VALUE					
ISUP Parameter	IAM; Access transport parameter HLC:	HLC_VALUE					
values:							
Comments:	SIP	SUT I	SUP				
	INVITE →	→ 1	AM				
	180 Ringing ←	← /	ACM				
	Ri	ging tone					
	200 OK INVITE ←	← /	ANM				
	ACK →						
	Co	nversation					
	BYE →	→ F	REL				
	200 OK BYE ←	← F	RLC				

	Values and selection criteria for the test purpose TP101022
VA_01	HLC_VALUE = Telephony
	BC_VALUE = speech
VA_02	HLC_VALUE = Facsimile Group 2/3 (ITU-T Recommendation F.182 [20])
	BC_VALUE = 3,1 kHz audio
VA_03	HLC_VALUE == Facsimile Group 4 Class I (ITU-T Recommendation F.184 [20])
	BC_VALUE = Unrestricted digital information
VA_04	HLC_VALUE == Teletex service, basic and mixed mode of operation
	(ITU-T Recommendation F.230 [22]) and facsimile service Group 4, Classes II and III
	(ITU-T Recommendation F.184 [21])
	BC_VALUE = Unrestricted digital information
VA_05	HLC_VALUE == Teletex service, basic and processable mode of operation
	(ITU-T Recommendation F.220 [23])
	BC_VALUE = Unrestricted digital information
VA_06	HLC_VALUE = Teletex service, basic mode of operation (ITU-T Recommendation F.200
	[24])
	BC_VALUE = Unrestricted digital information
VA_07	HLC_VALUE = Syntax based Videotex (ITU-T Recommendations F.300 [25] and T.102)
	BC_VALUE = Unrestricted digital information
VA_08	HLC_VALUE = International Videotex interworking via gateways or interworking units (ITU-
	T Recommendations F.300 [25] and T.101)
	BC_VALUE = Unrestricted digital information
VA_09	HLC_VALUE = Telex service (ITU-T Recommendation F.60 [26])
	BC_VALUE = Unrestricted digital information
VA_10	HLC_VALUE = Message Handling Systems (MHS) (X.400 - Series Recommendations)
	BC_VALUE = Unrestricted digital information
VA_11	HLC_VALUE = OSI application (X.200 - Series ITU-T Recommendations)
	BC_VALUE = Unrestricted digital information
VA_12	HLC_VALUE = Audio visual (ITU-T Recommendation F.721 [27])
	BC_VALUE = Unrestricted digital information

TP101023	SIP reference: RF	C 3261 [6]		SUP reference: 7 [1], clause 7.2.3.1.2.10		
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/					
SIP selection criteria:						
ISUP selection						
criteria:						
Test purpose:	Mapping of PSTN XML BearerCapability and HighLayerCompatibility in User Teleservice parameter Ensure that the SUT in the Idle state on receipt of a INVITE message with PSTN XML BearerCapability BC_VALUE and the PSTN XML HighLayerCompatibility HLC_VALUE					
	sends an IAM message wit PSTN XML HighLayerCor		vice parameter	containing the received		
SIP Parameter	INVITE					
values:	PSTN XML BearerCapabi PSTN XML HighLayerCor		ALUE			
ISUP Parameter	IAM; User teleservice para	meter				
values:						
Comments:	SIP	SL	IT	ISUP		
	INVITE	→	→	IAM		
	180 Ringing	←	←	ACM		
		Ringing	g tone			
	200 OK INVITE	←	←	ANM		
	ACK	→				
		Conver	sation			
	BYE	→	→	REL		
	200 OK BYE	-	+	RLC		

	Values and selection criteria for the test purpose TP101023
VA_01	HLC_VALUE = Telephony
	BC_VALUE = speech
VA_02	HLC_VALUE = Facsimile Group 2/3 (ITU-T Recommendation F.182 [20])
	BC_VALUE = 3,1 kHz audio
VA_03	HLC_VALUE == Facsimile Group 4 Class I (ITU-T Recommendation F.184 [21])
	BC_VALUE = Unrestricted digital information
VA_04	HLC_VALUE == Teletex service, basic and mixed mode of operation (ITU-T
	Recommendation F.230 [22]) and facsimile service Group 4, Classes II and III (ITU-T
	Recommendation F.184 [21])
	BC_VALUE = Unrestricted digital information
VA_05	HLC_VALUE == Teletex service, basic and processable mode of operation (ITU-T
	Recommendation F.220 [23])
	BC_VALUE = Unrestricted digital information
VA_06	HLC_VALUE = Teletex service, basic mode of operation (ITU-T Recommendation F.200
	[24])
	BC_VALUE = Unrestricted digital information
VA_07	HLC_VALUE = Syntax based Videotex (ITU-T Recommendations F.300 [25] and T.102)
	BC_VALUE = Unrestricted digital information
VA_08	HLC_VALUE = International Videotex interworking via gateways or interworking units
	(ITU-T Recommendations F.300 [25] and T.101)
	BC_VALUE = Unrestricted digital information
VA_09	HLC_VALUE = Telex service (ITU-T Recommendation F.60 [26])
	BC_VALUE = Unrestricted digital information
VA_10	HLC_VALUE = Message Handling Systems (MHS) (X.400 - Series Recommendations)
	BC_VALUE = Unrestricted digital information
VA_11	HLC_VALUE = OSI application (X.200 - Series Recommendations)
	BC_VALUE = Unrestricted digital information
VA_12	HLC_VALUE = Audio visual (ITU-T Recommendation F.721 [27])
	BC_VALUE = Unrestricted digital information

TP101024	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.2.10	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial A	Address message (IAM)/	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of PSTN XML LowLayerCompatib	pility in ATP LLC	
	Francisco that the CLIT is the Idle atota as year	paint of a INIVITE manager with a DCTN VAN	
		ceipt of a INVITE message with a PSTN XML	
	LowLayerCompatibility		
	and an IANA manager with the Access to	an an aut manager to a containing the DCTN VMI	
		ansport parameter containing the PSTN XML	
OID D	LowLayerCompatibility as received in the INVITE message.		
SIP Parameter	INVITE; PSTN XML LowLayer Compatibility: LLC_VALUE (PIXIT)		
values:			
ISUP Parameter	IAM; Access transport parameter LLC: LLC_VALUE (PIXIT)		
values:			
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Ring	ging tone	
	200 OK INVITE ←	← ANM	
	ACK →		
		versation	
	BYE →	→ REL	
	200 OK BYE	← RLC	

TP101025	SIP reference:	RFC 3261 [6]	I	SUP reference:
			ES 283 02	27 [1], clause 7.2.3.1.2.10
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/			
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Mapping of PSTN XML	ProgressIndicator in J	ATP PI	
			pt of a INVITE r	nessage with a valid PSTN
	XML ProgressIndicate	r		
	sends an IAM message with the Access transport parameter containing the PSTN XML			er containing the PSTN XML
	ProgressIndicator as r			
SIP Parameter	INVITE; PSTN XML ProgressIndicator: PI_VALUE			
values:				
ISUP Parameter	IAM; progress indicator PI_VALUE			
values:				
Comments:	SIP	SI	UT	ISUP
	INVITE	→	→	IAM
	180 Ringing	←	←	ACM
		Ringir	ng tone	
	200 OK INVITE	←	+	ANM
	ACK	→		
		Conve	rsation	
	BYE	→	→	REL
	200 OK BYE	É	÷	RLC

Values and selection criteria for the test purpose TP101025		
VA_01 PI_VALUE = Call is not end-to-end ISDN; further call progress information is available		
	in-band (# 1)	
VA_02	PI_VALUE = Originating access is non ISDN (#3)	

TP101026	SIP reference: R	FC 3261 [6]		ISUP reference: 127 [1], clause 7.2.3.1.2.9
TSS reference:	SIP-ISUP/Basic call/ Sen	nding of the Initial Add	dress message	(IAM)/
SIP selection criteria:			_	
ISUP selection criteria:	PICS 4/3			
Test purpose:	HOP counter derived from the Max-Forward header Ensure that the SUT the I-MGCF shall derive the Hop Counter parameter value from the Max-Forwards header field value by applying a factor. The Hop Counter for a given message should never increase and should decrease by at least 1 with each successive visit to an MGCF, regardless of interworking, and similarly for Max-Forwards in the SIP domain.			
SIP Parameter	Max-Forward header			
values:				
ISUP Parameter values:	IAM: Hop Counter para	meter value		
Comments:	SIP	SUT		ISUP
	INVITE	→	→	IAM
	180 Ringing Ringing tone	←	←	ACM
	200 OK INVITE ACK Conversation	← →	←	ANM
	BYE	→	→	REL
	200 OK BYE	(←	RLC
				should be large enough to xpected of a validly routed

SIP reference: RFC 3261 [6]	ISUP reference:	
	ES 283 027 [1], clause 7.2.3.1.2.1	
SIP-ISUP/Basic call/ Sending of the Initial	Address message (IAM)/	
NOT PICS 1/5		
Mapping of Request URI into called party	number "national number"	
, -	t) number". The country code is removed from	
ŭ		
INVITE: Request URI		
IAM: Called party number		
SIP	SUT ISUP	
INVITE →	→ IAM	
180 Ringing ←	← ACM	
Rir	ging tone	
200 OK INVITE ←	← ANM	
ACK →		
	nversation	
BYE →	→ RFL	
	← RLC	
	SIP-ISUP/Basic call/ Sending of the Initial NOT PICS 1/5 Mapping of Request URI into called party Analyse the information contained in recei If CC is country code of the network in whi Address indicator to "National (significan the numberstring. INVITE: Request URI IAM: Called party number SIP INVITE 180 Ringing Rin 200 OK INVITE ACK Cor	

TP101028	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.1
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initia	Address message (IAM)/
SIP selection criteria:		
ISUP selection criteria:	PICS 1/5	
Test purpose:	Mapping of Request URI into called party number "international number" Analyse the information contained in received Request URI E.164 address. If CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "International number".	
SIP Parameter values:	INVITE: Request URI	
ISUP Parameter values:	IAM: Called party number	
Comments:	200 OK INVITE ← ACK →	SUT ISUP → IAM ← ACM riging tone ← ANM NVersation → REL ← RLC

TP101029	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clause 7.2.3.1.2.1
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Ad	
SIP selection	Ĭ	V , /
criteria:		
ISUP selection		
criteria:		
Test purpose:	network number indicator	mber. Setting of Numbering plan and internal
	Ensure that the SUT on receipt of an INVITE contained in the userinfo component of the R Internal Network Number Indicator: routing	
	 Numbering plan Indicator: 001 ISDN (Te 	lephony) numbering plan.
SIP Parameter values:	INVITE: Request URI	
ISUP Parameter values:	IAM: Called party number	
Comments:	SIP	UT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringir	ng tone
	200 OK INVITE ←	← ANM
	ACK →	
		ersation
	BYE →	→ REL
	200 OK BYE ←	← RLC

TP101030	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.2.1	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initi	al Address message (IAM)/	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of number digits in the Reques	r URI into address signals in the Called party	
	number CC is the same as the MGCF is	located	
	Analyse the information contained in red		
	If CC is country code of the network in which the next hop terminates, then remove "+CC"		
	and use the remaining digits to fill the A	ddress signals.	
SIP Parameter	INVITE: Request URI		
values:			
ISUP Parameter	IAM: Called party number address sig	nals	
values:			
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	F	Ringing tone	
	200 OK INVITE ←	← ANM	
	ACK →		
		Conversation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP101031	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.1	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial A	Address message (IAM)/	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of number digits in the Requesr URI into address signals in the Called party number CC is not the same as the MGCF is located Analyse the information contained in received E.164 address. If CC is not the country code of the network in which the next hop terminates, then remove "+" and use the remaining digits to fill the Address signals.		
SIP Parameter	INVITE: Request URI		
values:	INVITE: Nequest ON		
ISUP Parameter values:	IAM: Called party number address signal	S	
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Ring	ging tone	
	200 OK INVITE ←	← ANM	
	ACK →		
	Con	versation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP101032	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.1
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Ad	ddress message (IAM)/
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Mapping of calling party category	
	Ensure that a cpc SIP_CPC parameter SIP_the "language" in the Accept-Contact SIP_L/parameter ISUP_CPC in the sent IAM	CPC received in the P-Asserted-Identity and ANG header is mapped into the calling party
SIP Parameter	INVITE: P-Asserted-Identity, Accept-Contact	
values:		
ISUP Parameter	IAM: Calling Party Category	
values:		
Comments:	SIP	UT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringi	ng tone
	200 OK INVITE ←	← ANM
	ACK →	
	Conv	ersation
	BYE →	→ REL
	200 OK BYE ←	← RLC

Values for test purposes TP101032			
SIP	_CPC	ISUP_CPC	
cpc received in a P-Asserted-Identity	Accept-Contact 'language' SIP_LANG	Sent Calling party's category	
operator	French	operator, language French	
operator	English	operator, language English	
operator	German	operator, language German	
operator	Russian	operator, language Russian	
operator	Spanish	operator, language Spanish	
ordinary		ordinary calling subscriber	
test		Test call	
payphone		Payphone	
cellular		mobile terminal located in the home PLMN	
cellular-roaming		mobile terminal located in a visited PLMN	
ieps		IEPS call marking for preferential call set up	

6.2.1.2 Overlap procedure at the I-MGCF

FFS

6.2.1.3 Sending of COT

TP103001	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.3	
TSS reference:	SIP-ISUP/Basic call/ COT		
SIP selection	PICS 4/4 AND PICS 4/5		
criteria:			
ISUP selection	PICS 1/5 AND PICS 4/1		
criteria:	OOT: A fine Picture of		
Test purpose:	COT is sent after precondition met		
	If the IAM has already been sent, the Continuity message shall be sent indicating		
	"continuity check successful", when all of the		
	- The requested preconditions (if any) in th		
	- A possible outstanding continuity check procedure is successfully performed on the		
	outgoing circuit.		
SIP Parameter	INVITE: Require: precondition		
values:	SDP a=curr:qos local none		
	a=curr:qos remote none a=des:qos mandatory local sen	drecy	
	a=des:qos mandatory local sen a=des:qos none remote sendre		
	μετικές του		
	183: Require: 100rel		
	SDP a=curr:qos local none		
	a=curr:qos remote none	dragu	
	a=des:qos mandatory local sen a=des:qos mandatory remote s		
	a=conf:gos remote sendrecv	endrecv	
	UPDATE:	UPDATE:	
	SDP a=curr:qos local sendrecv		
	a=curr:qos remote none	a=curr:qos remote none a=des:qos mandatory local sendrecv	
	a=des:qos mandatory local sen a=des:qos mandatory remote s		
	a=des.qos mandatory remote s	endrecv	
	200 OK UPDATE		
	SDP a=curr:qos local sendrecv		
	a=curr:qos remote sendrecv		
	a=des:qos mandatory local sendrecv		
ISUP Parameter	a=des:qos mandatory remote s IAM: "Continuity check performed on a previo		
values:	this circuit"	us circuit or Continuity check required on	
TaidO3.	COT continuity indicator: Continuity check such	ccessful:	
Comments:		JT ISUP	
	INVITE ->	→ IAM	
	183 Session Progress ←		
	PRACK →		
	200 OK PRACK ←		
	UPDATE →	→ COT	
	200 OK UPDATE ←		
	180 Ringing ←	← ACM	
	PRACK →		
	200 OK PRACK ←	g tono	
	_	g tone ← ANM	
	200 OK INVITE ← ACK →	← ANM	
		rsation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

6.2.1.4 Receipt of the Address Complete Message (ACM)

TP104001	SIP reference: R	FC 3261 [6]		SUP reference:
			ES 283 ()27 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Red	ceipt of the Address c	omplete messa	ge (ACM)/
SIP selection	NOT PICS 4/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Early ACM not interwork	ed		
	Ensure that the SUT on r		ssage where the	ne Called party status
	indicator is set to "no ind	ication":		
	 the ACM is not in 	nterworked.		
SIP Parameter				
values:				
ISUP Parameter	ACM Called party status: no indication;			
values:				
Comments:	SIP	SL	JT	ISUP
	INVITE	→	→	IAM
			←	ACM (no indication)
	200 OK INVITE	←	←	ANM
	ACK	→		
		Conver	sation	
	BYE	→	→	REL
	200 OK BYE	-	É	RLC

TP104002	SIP reference: RFC 3261 [6]	· · · · · · · · · · · · · · · · · · ·	SUP reference: 027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Ado	dress complete messa	age (ACM)/
SIP selection criteria:	NOT PICS 4/15	·	
ISUP selection			
criteria:			
Test purpose:	Early ACM not interworked. Announced	ment provided by the	terminating network
	Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "no indication" and during the establishment of the communication the PSTN/ISDN provides an announcement: • the ACM is not interworked.		
SIP Parameter			
values:			
ISUP Parameter values:	ACM Called party status: no indication;		
Comments:	SIP	SUT	ISUP
	INVITE →	→	IAM
		←	ACM (no indication)
	Tones	s or announcement	
	200 OK INVITE ←	←	ANM
	ACK →		
		Conversation	
	BYE →	→	REL
	200 OK BYE ←	+	RLC

TP104003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/		
SIP selection	PICS 4/15 AND PICS 2/1		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Early ACM terminating access ISDN and OBCI "inband info available" received. Sending of 183 containing a P-Early-Media header Ensure that SUT on receipt of an ACM message where the CPS indicator is set to "no		
	indication" the ISUP indicator is set to "ISUP is used all the way", the ISDN access indicator is set to "ISDN" and the OBCI with the in-band information is set to "Yes" and if the I-MGCF has received the P-Early-Media header in the INVITE request, and has not already sent a provisional response including a P-Early-Media header with parameters indicating authorization of early media, then the I-MGCF shall:		
	 send the 183 Session Progress responserly media. 	onse with a P-Early-Media header authorizing	
SIP Parameter	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8		
values:	183 Session Progress: P-Early-Media header		
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
_	OBCI in-band information: Yes		
Comments:	SIP SU		
	INVITE →	→ IAM	
	183 Session Progress ←	← ACM (no indication)	
	200 OK INVITE ← ACK →	← ANM	
	Conve		
	BYE -	→ REL	
	200 OK BYE ←	← RLC	

TP104004	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address c		
SIP selection	PICS 4/15 AND PICS 2/1	<u> </u>	
criteria:			
ISUP selection			
criteria:			
Test purpose:	Early ACM BCI "ISDN User Part not used all the way" received. Sending of 183 containing		
	a P-Early-Media header		
	Ensure that SUT on receipt of an ACM messa	ge where the CPS indicator is set to "no	
	indication", the ISUP indicator is set to "ISUP		
	and if the I-MGCF has received the P-Early-M		
	not already sent a provisional response includ		
	indicating authorization of early media, then the I-MGCF shall:		
	and the 100 Occion December with D. Forth Madie		
	 send the 183 Session Progress response with P-Early Media and a P-Early-Media header authorizing early media. 		
SIP Parameter	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8		
values:	183 Session Progress: P-Early-Media header		
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	ISUP indicator: ISUP is not used all the way"		
0	OBCI with the in-band information: Yes SIP SUT ISUP		
Comments:			
	INVITE → 183 Session Progress ←	→ IAM ← ACM (no indication)	
	183 Session Progress ←	← ACM (no indication)	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conver	sation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP104005	SIP reference: RFC 3261 [6]		eference:	
TSS reference:	ES 283 027 [1], clause 7.2.3.1.4A SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/			
SIP selection	PICS 2/1 AND PICS 4/18	oompioto moodago (/ te	, , , , , , , , , , , , , , , , , , ,	
criteria:	. 100 _7 . 7 12 . 100 10			
ISUP selection				
criteria:				
Test purpose:	Early ACM BCI "Terminating access non-ISDN" received. Sending of 183 containing a PSTN XML ProgressIndicator #2			
	Ensure that the SUT, on receipt of an ACM r "no indication", the ISUP indicator is set to "	ISUP is used all the wa		
	indicator is set to "non-ISDN", then the I-MC	CF shall:		
	a and the 192 Cassian Dragges room	anaa with DCTN VML I	Dragraaaladiaatar	
	send the 183 Session Progress resp body containing the progress description			
SIP Parameter	body containing the progress descriptions "destination address is non-ISDN (#2)". INVITE: P-Early-Media header, SDP audio xxxx RTP/AVP 8			
values:	183 Session Progress: PSTM XML ProgressIndicator			
ISUP Parameter	ACM: CPS indicator: no indication,			
values:	ISUP indicator: ISUP is used all the way			
		ISDN access indicator: non-ISDN		
	OBCI with the in-band information: No			
Comments:	SIP SUT	ISUI	,	
		- "	A / ' ' ' ' ' ' '	
	183 Session Progress	← ACN	(no indication)	
	200 OK INIVITE 4	∠ ∧NIN	A	
		▼ AININ	'I	
	1 1 2 1 2			
		→ RFI		
	1			
	INVITE 183 Session Progress 200 OK INVITE ACK Conversation BYE	→ IAM ← ACM ← ANM → REL	1 (no indication)	

TP104006	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address co	omplete message (ACM)/	
SIP selection	PICS 2/1 AND PICS 4/18	_ · · · · · · · · · · · · · · · · · · ·	
criteria:			
ISUP selection			
criteria:			
Test purpose:	Early ACM BCi "ISDN User Part not used all the way" received. Sending of 183 containing		
	a PSTN XML ProgressIndicator #1		
	Ensure that the SUT on receipt of an ACM me		
	indication", the ISUP indicator is set to "ISUP shall:	not used all the way, then the i-MGCF	
	 send the 183 Session Progress response PSTN XML body containing the progress descriptions "call is not end-to-end ISDN, further call progress information is available in-band (#1). 		
SIP Parameter	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8		
values:	183 Session Progress: P-Early Media and PSTN XML ProgressIndicator body containing		
	the progress descriptions "call is not end-to-end ISDN, further call progress information is		
	available in-band (#1)		
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	ISUP indicator: ISUP not used all the	,	
Comments:	SIP SU	IT ISUP → IAM	
	· · · · · · -	2	
	183 Session Progress ←	← ACM (no indication)	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conver	sation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP104007	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.4A	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address c	omplete message (ACM)/	
SIP selection	PICS 2/1 AND PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Early ACM OBCI "Inband-info available" received. Sending of 183 containing a PSTN XML ProgressIndicator #8		
	Ensure that the SUT on receipt of an ACM me indication", the OBCI is set to "Inband-info ava		
	 send the 183 Session Progress response PSTN XML body containing the progress descriptions " in-band information or an appropriate pattern is now available" (#8). 		
SIP Parameter	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8		
values:	183 Session Progress: P-Early Media and PSTN XML ProgressIndicator body containing		
	the progress descriptions "in-band information or an appropriate pattern is now available" (#8)		
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	OBCI: in-band information or an appropriate pattern is now available		
Comments:	SIP SU		
	INVITE →	→ IAM	
	183 Session Progress ←	 ACM (no indication) 	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conver		
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP104008	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/		
SIP selection	PICS 2/1 AND PICS 4/18		
criteria:			
ISUP selection criteria:			
Test purpose:	Early ACM terminating access ISDN and ATP with PI received. Sending of 183 containing a PSTN XML ProgressIndicator		
	Ensure that the SUT on receipt of an ACM me indication", the ISUP indicator is set to "ISUP indicator set to "ISDN", the Access Transpo	used all the way", the ISDN access	
	 indicator set to PI_VALUE: sends a 183 Session Progress message with PSTN XML ProgressIndicator set to PI_VALUE". 		
SIP Parameter	183 Session Progress message PSTN XML F	ProgressIndicator set to PI_VALUE,	
values:			
ISUP Parameter	ACM, CPS indicator: no indication (00)		
values:	Called party"s category indicator: no indication(00) or ordinary subscriber (01) or		
	payphone (10) interworking indicator: no interworking encountered (0)		
	ISUP indicator: ISUP used all the way		
	ISDN access indicator: ISDN		
	ATP progress indicator: PI_VALUE		
	access delivery information: Set-up message	ge generated (IF PRESENT)	
Comments:	SIP SI		
	INVITE →	→ IAM	
	183 Session Progress ←	← ACM (no indication, ATP)	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conve	rsation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP104009	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.4A
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/		
SIP selection	PICS 2/1 AND PICS 4/18	•	,
criteria:			
ISUP selection			
criteria:			
Test purpose:	Early ACM received, mapping of BCI into PST	N XML Progre	ssIndicator #7 in the sent 183
SIP Parameter values: ISUP Parameter values:	Ensure that the SUT on receipt of an ACM message where the CPS indicator is set to "no indication", the interworking indicator I set to no interworking encountered, the ISUP indicator is set to "ISUP is used all the way", the ISDN access indicator is set to "ISDN" and the: • send the 183 Session Progress response with PSTN XML ProgressIndicator the progress descriptions "Terminating access ISDN" (#7). 183 Session Progress message PSTN XML ProgressIndicator set to value #7 ACM: CPS indicator: no indication interworking indicator: no interworking encountered		
	ISUP indicator: ISUP used all the way		
	ISDN access indicator: terminating access Is	SDN	
Comments:	SIP SUT	_	ISUP
	INVITE -	→	IAM
	183 Session Progress ←	←	ACM (no indication)
	200 OK INVITE ← ACK → Conversation	←	ANM
	BYE →	→	REL
	200 OK BYE ←	-	RLC

TP104010	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address	complete message (ACM)/
SIP selection criteria:		
ISUP selection criteria:		
Test purpose: SIP Parameter values:	ISDN access indicator set to ISDN_ACCES_OBCI_INBAND then:	ISUP indicator parameter set to ISUP_ID , the
ISUP Parameter values:	ACM FCI: ISUP_ID, ISDN_ACCESS_ID OBCI: OBCI_INBAND:	
Comments:	, , , , , , , , , , , , , , , , , , , ,	UT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringir	ng tone
	200 OK INVITE ← ACK →	← ANM
	Conve	ersation
	BYE →	→ REL
	200 OK BYE	← RLC

Table 3

test purposes	ISUP Parameter values:
VA_01	ACM ISUP_ID: ISUP not used all the way
	OBCI_INBAND: no
VA_02	ACM
	ISUP_ID: ISUP not used all the way
	OBCI_INBAND: yes
VA_03	ACM
	ISUP_ID: ISUP used all the way
	ISDN_ACCES_ID: non ISDN
	OBCI_INBAND: no
VA_04	ACM
	ISUP_ID: ISUP used all the way
	ISDN_ACCES_ID: non ISDN
	OBCI_INBAND: yes
VA_05	ACM
	ISUP_ID: ISUP used all the way
	ISDN access indicator: ISDN
	OBCI_INBAND: yes

TP104011	SIP reference: RFC 3261 [6]	ISUP reference:
11 104011	on reference. Ki o 3201 [o]	ES 283 027 [1], clause 7.2.3.1.4
		ETS 300 356-1 [28] clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address of	
SIP selection	PICS 2/1 AND PICS 4/18	omplete message (//ow///
criteria:	1 100 2/1 AND 1 100 4/10	
ISUP selection		
criteria:		
Test purpose:	ACM subscriber free received, mapping of Bo	CI into PSTN XML ProgressIndicator #7 in
	the sent 180	
	Ensure that the SUT on receipt of an ACM me	
	"subscriber free", the ISUP indicator is set to	"ISUP is used all the way", the ISDN access
	indicator is set to "ISDN":	
	sends an 180 Ringing response with I Transfer time.	
OID D	ProgressIndicator "Terminating acce	, ,
SIP Parameter	180 Ringing response with P-Early Media and	PSIN XML ProgressIndicator
values:	"Terminating access ISDN"(#7).	
ISUP Parameter	ACM; CPS indicator: subscriber free	
values:	ISUP indicator: ISUP is used all the way,	
0	ISDN access indicator: ISDN	IT IOLID
Comments:	SIP SU	
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringin	
	200 OK INVITE ←	← ANM
	ACK →	
	Conver	sation
	BYE →	→ REL
	200 OK BYE ←	← RLC

TP104012	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address c	
SIP selection criteria:	PICS 2/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	descriptions set to PI_ID. 180 Ringing	M message, where the CPS indicator is set parameter set to ISUP_ID, the ISDN d the OBCI in-band information set to PSTN XML body containing the progress
values:	P-Early Media and PSTN XML ProgressIndicator set to PI_ID.	
ISUP Parameter	ACM; CPS indicator: subscriber free	
values:	ISUP indicator: ISUP_ID ISDN access indicator: ISDN_ACCES_ID	
Comments:	SIP SL	JT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringing	
	200 OK INVITE ← ACK →	← ANM
	Conver	rsation
	BYE →	→ REL
	200 OK BYE ←	← RLC

Values for test purposes TP104012		
test purposes	ISUP Parameter values:	PSTN XML progress descriptions:
VA_01	ISUP_ID: ISUP not used all the way OBCI_INBAND: no	PI_ID: Call is not end-to-end ISDN (#1)
VA_02	ISUP_ID: ISUP not used all the way OBCI_INBAND: yes	PI_ID: Call is not end-to-end ISDN (#1) and In-band information or appropriate pattern now available (#8)
VA_03	ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: no	PI_ID: Destination address is non-ISDN (#2)
VA_04	ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: yes	PI_ID: Destination address is non-ISDN (#2) and In-band information or appropriate pattern now available (#8)
VA_05	ISUP_ID: ISUP used all the way ISDN access indicator: ISDN OBCI_INBAND: yes	PI_ID: In-band information or appropriate pattern now available (#8) and <i>Terminating access ISDN"(#7)</i>

TP104013	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4.1 ETS 300 356-1 [28] clauses 2.1.4, 2.2
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address of	
SIP selection	PICS 4/18	
criteria:		
ISUP selection criteria:	PICS 4/19	
Test purpose:	ACM received containing a TMU parameter a ProgressIndicators in the 180 Ensure that the SUT on receipt of an ACM me "subscriber free", the ISUP indicator is set to 'indicator is set to "ISDN" and the Transmissivalue ISDN_BC_VALUE and the Access TraATP_VALUE: • sends an 180 Ringing message with a	essage where the CPS indicator is set to "ISUP is used all the way", the ISDN access on Medium Used (TMU) is included with the
		gressIndicator body containing the progress
SIP Parameter values:	INVITE; PSTN XML first Bearer Capability: INVITE PSTN XML second Bearer Capability: INVIT	
10110 0	180 Ringing PSTN XML BC: BC_VALUE	
ISUP Parameter	ACM; CPS indicator: subscriber free,	
values:	ISUP indicator: ISUP is used all the way	
	ISDN access indicator: ISDN TMU: TMU_VALUE	
	ATP: BC ATP_VALUE	
Comments:		JT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringin	a tone
	200 OK INVITE ←	← ANM
	ACK →	
	Conve	rsation
	BYE →	→ REL
	200 OK BYE ←	← RLC

Values and selection criteria for test purpose TP104013			
Test	ACM Parameter values	180 Ringing Parameter	INVITE parameter value
purposes		values:	
VA_01	TMU_VALUE: speech ATP_VALUE: no BC	PSTN XML BC_VALUE: speech	PSTN XML INVITE BC1: speech INVITE BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: 3,1 kHz ATP_VALUE: no BC	PSTN XML: BC_VALUE: 3,1 kHz audio	PSTN XML INVITE BC1: 3,1 kHz audio INVITE BC2: unrestricted digital information with tones and announcements
VA_03	TMU_VALUE: speech ATP_VALUE: BC = 3,1 kHz audio	PSTN XML BC_VALUE: 3,1 kHz audio	PSTN XML INVITE BC1: 3,1 kHz audio INVITE BC2: unrestricted digital information with tones and announcements
VA_04	TMU_VALUE: speech ATP_VALUE: BC = speech	PSTN XML BC_VALUE: speech	PSTN XML INVITE BC1: speech INVITE BC2: unrestricted digital information with tones and announcements
VA_05	TMU_VALUE: 3,1 kHz ATP_VALUE: BC = speech	PSTN XML BC_VALUE: speech	PSTN XML INVITE BC1: speech INVITE BC2: unrestricted digital information with tones and announcements
VA_06	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC = 3,1 kHz audio	PSTN XML BC_VALUE: 3,1 kHz audio	PSTN XML INVITE BC1: 3,1 kHz audio INVITE BC2: unrestricted digital information with tones and announcements

TP104014	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4.1 ETS 300 356-1 [28] clause 2.1.4	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address of	complete message (ACM)/	
SIP selection	PICS 4/18		
criteria:			
ISUP selection criteria:	PICS 4/19		
Test purpose:	Fallback occurs in the early ACM mapping of TMU parameter and BC in ATP into 183 PSTN XML ProgressIndicators Ensure that the SUT on receipt of an ACM message where the CPS indicator is set to "no		
	indication", the ISUP indicator is set to "ISUP indicator is set to "ISDN" and the Transmissio value TMU_VALUE and theBC in the Access ATP_VALUE:	n Medium Used (TMU) is included with the Transport Parameter (ATP) set to	
	ISDN"(#7).		
SIP Parameter	INVITE;		
values:	PSTN XML first Bearer Capability: INVITE_BC1		
	PSTN XML second Bearer Capability: INVII		
ISUP Parameter	183 Session Progress; PSTN XML BearerCa ACM; CPS indicator: no indication.	Pability. ISDN_BC_VALUE	
values:	ISUP indicator: ISUP is used all the way		
values.	ISDN access indicator: ISDN		
	TMU: TMU_VALUE		
	ATP: BC ATP_VALUE		
Comments:	SIP SL	JT ISUP	
	INVITE →	→ IAM	
	183 Session Progress	← ACM	
	180 Ringing ←	← CPG	
	Ringin	~	
	200 OK INVITE	← ANM	
	ACK →		
	Conve		
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

	Values and selection criteria for test purpose TP104014			
Test purposes	ACM Parameter values	183 Session Progress Parameter values:	INVITE parameter value	
VA_01	TMU_VALUE: speech ATP_VALUE: no BC	PSTN XML BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements	
VA_02	TMU_VALUE: 3,1 kHz ATP_VALUE: no BC	PSTN XML BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements	
VA_03	TMU_VALUE: speech ATP_VALUE: BC = 3,1 kHz	PSTN XML BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements	
VA_04	TMU_VALUE: speech ATP_VALUE: BC = speech	PSTN XML BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements	
VA_05	TMU_VALUE: 3,1 kHz ATP_VALUE: BC = speech	PSTN XML BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements	
VA_06	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC = 3,1 kHz audio	PSTN XML BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements	

TP104015	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.	
		ETS 300 356-1 [28] clause 7.	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/		
SIP selection	PICS 4/18		
criteria:			
ISUP selection	PICS 4/19		
criteria:	F-IIIIIII	U.O. and DO in ATD into 400 DO	OTAL VALU
Test purpose:	Fallback occurs in the early ACM mapping of ProgressIndicators	ALC una BC IN ATP Into 183 PS	STIN XIVIL
	Ensure that the SUT on receipt of an ACM message where the CPS indicator is set to "no indication", the ISUP indicator is set to "ISUP is used all the way", the ISDN access indicator is set to "ISDN" and containing an Access Transport Parameter (ATP) including a High Layer Compatibility (HLC) and containing the progress indicator #5: "interworking has occurred and has resulted in a telecommunication service change":		
	sends a 183 Session Progress messa the progress indication "interworking I telecommunication service change" (# the PSTN XML HighLayerCapability	ge with PSTN XML ProgressI n as occurred and has resulted ir 5) " <i>Terminating access ISDN"(</i> a	idicator with
SIP Parameter	INVITE: HLC : HLC_VALUE1 (PIXIT), HLC_V		
values:	180 Session Progress; PSTN XML ProgressIndicator: interworking has occurred and has		
	resulted in a telecommunication service changement PSTN XML HighLayerCapability: HLC_VAL		
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
	ATP: progress indicator: interworking has o	ccurred and has resulted in a	
	telecommunication service change (#5) HLC : HLC_VALUE2 (PIXIT)		
Comments:	SIP SU	T ISUP	
	INVITE →	→ IAM	
	183 Session Progress	← ACM	
	180 Ringing ←	← CPG	
	Ringin	g tone	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conve	sation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP104016	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address	complete message (ACM)/	
SIP selection	PICS 4/18		
criteria:			
ISUP selection criteria:	PICS 4/19		
Test purpose:	ACM received, mapping of ATP HLC and PI	#5 into 180 PSTN XML ProgressIndicators	
	Ensure that the SUT on receipt of an ACM message where the CPS indicator is set to "subscriber free", the ISUP indicator is set to "ISUP is used all the way", the ISDN access indicator is set to "ISDN" and containing an Access Transport Parameter (ATP) including a High Layer Compatibility (HLC) and containing the progress indicator #5: "interworking has occurred and has resulted in a telecommunication service change":		
	progress indication "interworking has telecommunication service change" (the PSTN XML HighLayerCapability	#5) "Terminating access ISDN"(#7) and with	
SIP Parameter	INVITE: PSTN XML HighLayerCapability: H	ILC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT)	
values:	180 Ringing;		
	PSTN XML ProgressIndicator: interworking has occurred and has resulted in a telecommunication service change (#5)		
	PSTN XML HighLayerCapability.: HLC_VAI	LIE2 (DIVIT)	
ISUP Parameter	ACM; CPS indicator: subscriber free,	LOEZ (FIXIT)	
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
	ATP: progress indicator: interworking has of	occurred and has resulted in a	
	telecommunication service change (#5)		
	HLC: HLC_VALUE2 (PIXIT)		
Comments:		UT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		ng tone	
	200 OK INVITE	← ANM	
	ACK →		
		ersation	
	BYE -	→ REL	
	200 OK BYE ←	← RLC	

TP104017	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address c	omplete message (ACM)/
SIP selection criteria:	PICS 4/15 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	ACM free terminating access ISDN received. Sending of 180 containing a P-Early-Media header Ensure that SUT on receipt of an ACM message where the CPS indicator is set to	
	"subscriber free" and if the I-MGCF has received request, and has not already sent a provisional with parameters indicating authorization of each send the 180 Ringing response with media	ved the P-Early-Media header in the INVITE al response including a P-Early-Media header rly media, then the I-MGCF shall: a P-Early-Media header authorizing early
SIP Parameter values:	INVITE: P-Early-Media header, SDP audio xx 180 Ringing: P-Early-Media header	xxx RTP/AVP 8
ISUP Parameter values:	ACM; CPS indicator: free,	
Comments:	SIP SL	JT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringin	-
	200 OK INVITE ← →	← ANM
	Conver	
	BYE →	→ REL
	200 OK BYE ←	← RLC

TP104018	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address	
SIP selection	PICS 4/18	
criteria:		
ISUP selection criteria:		
Test purpose:	ACM free received contains an ATP convey in the sent 180	ing the LLC, mapping into the PSTN XML LLC
	ATP set to LLC_VALUE:	nessage containing the LLC parameter in the
	 sends 180 response with a PSTN 2 element set to LLC_VALUE 	KML LowLayerCompatibility information
SIP Parameter values:	180: PSTN XML LowLayerCompatibility:	LC_VALUE (PIXIT)
ISUP Parameter	ACM; ATP LLC: LLC_VALUE	
values:		
Comments:		SUT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	_	ing tone
	200 OK INVITE ←	← ANM
	ACK →	_
		ersation
	BYE →	→ REL
	200 OK BYE ←	← RLC

TP104019	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Addre	ss complete message (ACM)/
SIP selection	PICS 4/18	
criteria:		
ISUP selection criteria:		
Test purpose:	XML LLC in the sent 183	TP conveying the LLC, mapping into the PSTN I message containing the LLC parameter in the
	sends 183 Session Progress res LowLayerCompatibility information	ponse with a PSTN XML tion element set to LLC_VALUE
SIP Parameter	180: PSTN XML LowLayerCompatibility	: LLC_VALUE (PIXIT)
values:		
ISUP Parameter	ACM; ATP LLC : LLC_VALUE	
values:		
Comments:	SIP	SUT ISUP
	INVITE →	→ IAM
	183 Session Progress ←	← ACM
	180 Ringing ←	← CPG
	Rii	nging tone
	200 OK INVITE ←	← ANM
	ACK →	
	Co	nversation
	BYE →	→ REL
	200 OK BYE ←	← RLC

6.2.1.5 Receipt of the Call progress message (CPG)

TP105001	SIP reference: RFC 32	261 [6]		ISUP reference:
			ES 283	027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of	of the Call progr	ess message (CPG).
SIP selection	NOT PICS 4/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	CPG Alerting is mapped into a	180 ringing		
	Ensure that the SUT, having re			
	indication", on receipt of a CPC	message wne	ere the event ir	itormation parameter event
	indicator is set to "Alerting":			
	the 190 Pinging SIP re	enonco ic cont		
SIP Parameter	the 180 Ringing SIP re	sponse is sem	•	
values:				
ISUP Parameter	ACM: Called party status	"no indication"		
values:	CPG; event information		ent indicator: A	Jertina
Comments:	SIP	•	UT	ISUP
	INVITE	→	→	IAM
			←	ACM (no indication)
	180 Ringing	←	+	CPG (Alerting)
	100 Kinging	=	ng tone	or o (Alerting)
	200 OK INVITE	←	tg torie	ANM
	ACK	→	•	ALVIVI
	AON	-	ersation	
	BYE	→ Conve	→ •	REL
	200 OK BYE	-	→	RLC
	ZUU UN BTE			NLU

TP105002	SIP reference: RFC	3261 [6]	ES 283	ISUP reference: 027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt	of the Call pro	gress message (CPG).
SIP selection criteria:	NOT PICS 4/15			
ISUP selection criteria:				
Test purpose:	indicator is set to "Progress"	received the APG message v		lled party status indicator "no nformation parameter event
	the CPG is not interw	orked.		
SIP Parameter				
values:				
ISUP Parameter	ACM: Called party status "no			
values:	CPG; event information pa	rameter ever		
Comments:	SIP		SUT	ISUP
	INVITE	→	→	IAM
			+	ACM (no indication)
			←	CPG (Progress)
	200 OK INVITE ACK	← →	←	ANM
	5) (5		nversation	DE!
	BYE	→	→	REL
	200 OK BYE	<u> </u>	<u> </u>	RLC

TP105003	SIP reference: RFC 3261 [6]	ES 283	ISUP reference: 027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call p	rogress message (CPG).
SIP selection	NOT PICS 4/15		
criteria:			
ISUP selection			
criteria:			
Test purpose:	CPG "in-band information or an appropria	te pattern is now a	vailable" is not interworked
	Ensure that the SUT, having received the		
	indication", on receipt of a CPG message	where the event ii	ntormation parameter event
	indicator is set to "in-band information or	ап арргорпате ра	llerri is now available .
	the CPG is not interworked.		
SIP Parameter	the CPG is not interworked.		
values:			
ISUP Parameter	ACM: Called party status "no indication"		
values:	CPG; event information parameter ever	nt indicator: in-hai	nd-information or an appropriate
values.	pattern is now available	it indicator. in bai	nd information of an appropriate
Comments:	SIP	SUT	ISUP
	INVITE -	→	IAM
		-	ACM (no indication)
		_	riom (no maisanom)
		←	CPG (Inband info)
		_	or o (meana ime)
	200 OK INVITE ←	←	ANM
	ACK →		,
	1, 1-11	onversation	
	BYE →	→	REL
	200 OK BYE ←	<u> </u>	RLC
l		_	_

TP105004	SIP reference: RFC	3261 [6]	ES		ISUP reference: 027 [1], clause 7.2.3.1.4A
TSS reference:	SIP-ISUP/Basic call/ Receipt	t of the Cal			
SIP selection	PICS 4/15		<u> </u>		/
criteria:					
ISUP selection					
criteria:					
Test purpose:	CPG "in-band information or	an approp	riate pattern is l	now a	vailable" is interworked, a P-
	Early-Media header is sent				
	Ensure that the SUT, having				
					formation parameter event
	indicator is set to "in-band in	nionnation	ог ап арргорпа	ne pan	erri is now available .
	a 192 Consign Brogn	raca ia cant	containing the	D Earl	v Madia Haadar
SIP Parameter	 a 183 Session Progr 183 Session Progress: P-Ea 			r-Laii	y-iviedia i leader.
values:	103 Session Flogress. F-La	ny-ivicula i	leadel		
ISUP Parameter	ACM: Called party status "n	o indication	า"		
values:				in-ban	d-information or an appropriate
	pattern is now available				
Comments:	SIP		SUT		ISUP
	INVITE	→		→	IAM
				←	ACM (no indication)
					,
	183 Session Progress	←		←	CPG (Inband info)
					,
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	→		→	REL
	200 OK BYE	←		←	RLC

TP105005	SIP reference: RFC	C 3261 [6]			ISUP reference:
					027 [1], clause 7.2.3.1.4
					0 356-1 [28] clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Recei		II progress mes	sage (CPG).
SIP selection	PICS 4/15 AND PICS 4/18				
criteria:					
ISUP selection					
criteria:	000 11 11 1 100		55 / 14 //		
Test purpose:	CPG Alerting received, 180) containing	a P-Early-Medi	a head	der is sent
	Ensure that the SUT, havin	a received t	he ACM messa	ge. on	receipt of a CPG message
	· · · · · · · · · · · · · · · · · · ·	•		J ,	set to "Alerting" without BCI
	included:	•			G
	sends an 180 Ringi	ing respons	e with P-Early N	/ledia	
SIP Parameter	180 Ringing PSTN XML Pro	ogressIndi	cator PI_ID, P-I	Early-I	Media
values:					
ISUP Parameter	CPG; event information p	CPG; event information parameter event indicator: Alerting			
values:					
Comments:	SIP		SUT		ISUP
	INVITE	→		→	IAM
	183 Session Progress	←		+	ACM (no indication)
	180 Ringing	←		←	CPG (Alerting)
		_	Ringing tone	_	o. o (g)
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	→		→	REL
	200 OK BYE	-		<u></u>	RLC

TP105006	SIP reference: I	RFC 3261 [6]		ISUP reference:
				027 [1], clause 7.2.3.1.4
				0 356-1 [28] clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Re	eceipt of the Call prog	ress message (CPG).
SIP selection	PICS 4/18			
criteria:				
ISUP selection criteria:				
Test purpose:	CPG Alerting contains a	an ATP with Progress	Indicator	
	Ensure that SUT having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "Alerting" and the ATP containing the progress indicator PI_VALUE: • sends an 180 Ringing response with PSTN XML body containing containing PSTN XML ProgressIndicator set to PI_VALUE.			
SIP Parameter	180 Ringing PSTN XML progress indicator PLID			
values:	3 3 3	1 3	_	
ISUP Parameter values:	CPG; event informatio	n parameter event i	ndicator: Alerti	ng
Comments:	SIP		SUT	ISUP
	INVITE	→	→	IAM
			←	ACM (no indication)
	180 Ringing	← Ring	ng tone	CPG (Alerting)
	200 OK INVITE	+	←	ANM
	ACK	→		
		Conv	ersation	
	BYE	→	→	REL
	200 OK BYE	←	←	RLC

TP105007	SIP reference: RFC 3261	[6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4	
TSS reference:	SIP-ISUP/Basic call/ Receipt of th	e Call progre	ress message (CPG).	
SIP selection criteria:	PICS 4/18			
ISUP selection criteria:				
Test purpose:	CPG Alerting containing a BCl and OBCl, mapping into PSTN XML instance Ensure that the SUT having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "Alerting" the ISUP indicator parameter set to ISUP_ID, the ISDN access indicator set to ISDN_ACCES_ID and the OBCl in-band information set to OBCl_INBAND: • sends an 180 Ringing response with P-Early Media and PSTN XML body containing with the progress indicator information element set to PI_ID.			
values:	180 Ringing PSTN XML progres	s indicator i	H_ID	
ISUP Parameter	CPG; event information parame	tor event in	adicator: Alorting	
values:	or o, event information parame	ter event in	idicator. Alerting	
Comments:	SIP INVITE →		UT ISUP → IAM ← ACM (no indication)	
	180 Ringing	Ringin	← CPG (Alerting) ng tone ← ANM	
	200 OK INVITE ← ACK →		← ANM ersation	
	BYE → 200 OK BYE ←		→ REL ← RLC	

	Values for test purposes	TP105007
test purposes	ISUP Parameter values:	PSTN XML progress descriptions:
VA_01	CPG ISUP_ID: ISUP not used all the way	PI_ID: Call is not end-to-end ISDN (#1)
VA_02	ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: no	PI_ID: Destination address is non-ISDN (#2)
VA_03	ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: yes	PI_ID: Destination address is non-ISDN (#2) and In-band information or appropriate pattern now available (#8)
VA_04	ISUP_ID: ISUP used all the way ISDN access indicator: ISDN	PI_ID: ("Terminating access ISDN"(#7))0.
VA_05	ISUP_ID: ISUP used all the way ISDN access indicator: ISDN OBCI_INBAND: yes	PI_ID: In-band information or appropriate pattern now available (#8) and (" <i>Terminating access ISDN</i> "(#7))

TP105008	SIP reference: RFC 3	261 [6]		ISUP reference: 027 [1], clause 7.2.3.1.4A	
TSS reference:	SIP-ISUP/Basic call/ Receipt of	of the Call prog		0 356-1 [28] clause 2.1.4	
SIP selection	PICS 4/18	n the Call plog	iess illessage (Ci G).	
criteria:	1 100 4/10				
ISUP selection criteria:					
Test purpose:	CPG Progress containing a Bo	CI and OBCI, n	napping into PS	TN XML instance	
SIP Parameter values:	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "Progress" and the ISUP indicator parameter set to ISUP_ID, the ISDN access indicator set to ISDN_ACCESS_ID and the OBCI in-band information set to OBCI_INBAND: • sends a 183 Session Progress with PSTN XML body containing the progress indicator set to PI_ID. 183 Session Progress PSTN XML progress indicator PI_ID				
ISUP Parameter	CPG; event information parameter event indicator: Progress				
values:	ISUP indicator: CPG_ISUP_I		•		
	ISDN access indicator: CPG	_ISDN_ACCES	SS_ID		
Comments:	SIP	_	SUT	ISUP	
	INVITE	→	→	IAM	
	183 Session Progress	←	+	ACM (no indication)	
	183 Session Progress	← Ringi	← ng tone	CPG (Progress)	
	200 OK INVITE	←	←	ANM	
	ACK	→			
		Conv	ersation		
	BYE	→	→	REL	
	200 OK BYE		+	RLC	

	Values for test purposes TP105008				
Test purposes	ISUP Parameter values:	ISDN Parameter values:			
VA_01	CPG CPG_ISUP_ID: ISUP not used all the way	PI_ID: Call is not end-to-end ISDN (#1)			
VA_02	CPG CPG_ISUP_ID: ISUP used all the way CPG_ISDN_ACCES_ID: non ISDN	PI_ID: Destination address is non-ISDN (#2)			
VA_03	CPG CPG_ISUP_ID: ISUP used all the way CPG_ISDN access indicator: ISDN	PI_ID: ("Terminating access ISDN"(#7).			

TP105009	SIP reference: RFC	3261 [6]		ISUP reference:	
)27 [1], clause 7.2.3.1.4A	
				0 356-1 [28] clause 2.1.4	
TSS reference:	SIP-ISUP/Basic call/ Receipt	of the Call progr	ess message (CPG).	
SIP selection	PICS 4/15 AND PICS 4/18				
criteria:					
ISUP selection criteria:					
Test purpose:	CDC in hondinformation or			silahla santaining a DOI and	
rest purpose.	OBCI, mapping into PSTN X	ML instance and	P-Early-Media		
	an appropriate pattern is nov ISDN access indicator set to and if the I-MGCF has receiv MGCF shall:	n parameter eve v available", the I- CPG_ISDN_ACO red the P-Early-M Progress with P-I	nt indicator is SUP indicator s CESS_ID ledia header in	set to "In-band information or	
SIP Parameter		183 Session Progress with P-Early Media and PSTN XML ProgressIndicator: PI_ID			
values:	Too Coocier Fregress marr	Larry Modia arre		rogrocomandator: r i_rb	
ISUP Parameter	CPG; event information pa	rameter event in	dicator: In-bar	nd information or an appropriate	
values:	pattern is now available				
	ISUP indicator : CPG_ISUP				
	ISDN access indicator: CP	G_ISDN_ACCES	_ID		
Comments:	SIP	S	UT	ISUP	
	INVITE	→	→	IAM	
			+	ACM (no indication)	
	183 Session Progress	← Ringir	← ng tone	CPG (In-band info)	
	200 OK INVITE	←	+	ANM	
	ACK	→			
		Conve	ersation		
	BYE	→	→	REL	
	200 OK BYE		←	RLC	

Values for test purposes TP105009				
test purposes	ISUP Parameter values:	ISDN Parameter values:		
VA_01	CPG CPG_ISUP_ID: ISUP used all the way CPG_ISDN_ACCES_ID: ISDN	PI_ID: In-band information or appropriate pattern now available (#8) and "Terminating access ISDN"(#7).		
VA_02	CPG_ISUP_ID: ISUP used all the way CPG_ISDN_ACCES_ID: non-ISDN	PI_ID: In-band information or appropriate pattern now available (#8) and Destination address is non-ISDN (#2)		
VA_03	CPG_ISUP_ID: ISUP not used all the way	PI_ID: In-band information or appropriate pattern now available (#8) and Call is not end-to-end ISDN (#1)		

TP105010	SIP reference: RFC 3261 [6]	ES 283 02 ETS 300	ISUP reference: 27 [1], clause 7.2.3.2.1.4.1 0 356-1 [28] clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call pro	ogress message (0	CPG).
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:	PICS 4/19		
Test purpose:	Fallback procedure: CPG Alerting received, mapping of TMU and BC in the included ATP into 180		
	Ensure that the SUT, having received the where the event information parameter event transmission medium used is included with transport Parameter is set to ATP_VALUE sends an 180 Ringing message with ISDN_BC_VALUE.	nt indicator is set n the value TMU_\ :	to " Alerting" and the VALUE and the Access
SIP Parameter	INVITE:		
values:	first Bearer Capability: INVITE_BC1 second Bearer Capability: INVITE_BC2 180 Ringing; PSTN XML BearerCapability: ISDN_TMU_	VALUE	
ISUP Parameter	CPG; event information parameter event indicator: Alerting		
values:	TMU: TMU_VALUE ATP: BC ATP_VALUE		
Comments:	SIP	SUT	ISUP
	INVITE →	→	IAM
		+	ACM (no indication)
	180 Ringing ←	←	CPG (Alerting)
	200 OK INVITE	ging tone	ANM
	ACK →		
		nversation	
	BYE →	→	REL
	200 OK BYE ←	<u></u>	RLC

TP105011	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.2.1.4.1	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).		
SIP selection	PICS 4/18		
criteria:			
ISUP selection	PICS 4/19		
criteria:			
Test purpose:	Fallback procedure: CPG Progress received into a 183 Ensure that the SUT, having received the AC	I, mapping of TMU and BC in the included ATP	
	where the event information parameter ev		
	Transmission medium requirement is incli		
	Access Transport Parameter is set to ATP		
	 sends a 183 Session Progress mess 	sage with the PSTN XML BearerCapability	
	encoded ISDN_BC_VALUE.		
SIP Parameter	INVITE;		
values:	first BearerCapability: INVITE_BC1		
	second BearerCapability: INVITE_BC2		
	180 Session Progress;		
	PSTN XML BearerCapability: ISDN_BC_V		
ISUP Parameter	CPG; event information parameter event indicator: Progress		
values:	TMU: TMU_VALUE		
	ATP: BC: ATP_VALUE		
Comments:	- · ·	SUT ISUP	
	INVITE ->	→ IAM	
		← ACM (no indication)	
	183 Session Progress ←	← CPG (Progress)	
	Ring	ing tone	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conv	versation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

	Values and selection c	riteria for test purposes TP1	05010 TP105011
Test purposes	ACM Parameter values	180 Ringing Parameter values:	INVITE parameter value
VA_01	TMU_VALUE: speech ATP_VALUE: no BC	ISDN_BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: 3,1 kHz ATP_VALUE: no BC	ISDN_BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements
VA_03	TMU_VALUE: speech ATP_VALUE: BC = 3,1 kHz	ISDN_BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements
VA_04	TMU_VALUE: speech ATP_VALUE: BC = speech	ISDN_BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_05	TMU_VALUE: 3,1 kHz ATP_VALUE: BC = speech	ISDN_BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_06	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC = 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements

TP105012	SIP reference: RFC 3261 [6	6]	ES 202 0	ISUP reference: 27 [1], clause 7.2.3.2.1.4.1
TSS reference:	SIP-ISUP/Basic call/ Receipt of the	Call progr		
SIP selection	PICS 4/18	Call plogie	ess message (CFG).
criteria:	1100 4/10			
ISUP selection				
criteria:				
Test purpose:	CPG Alerting received, sending of r and HLC in the 180	eceived Pl	#5 and HLC ii	n an ATP into a PSTN XML PI
	Ensure that the SUT, having receive where the event information parar an Access Transport Parameter in containing the progress indicator telecommunication service change" • sends an 180 Ringing mess progress indication "interwo	neter ever cluding a H #5: "interw : : :age with a :rking has o	nt indicator is ligh Layer Colorking has occ PSTN XML Poccurred and h	set to " Alerting" and containing mpatibility (HLC) and urred and has resulted in a rogressIndicator with the has resulted in a
	telecommunication service HighLayerCapability.	change" (#	5) and with the	PSTN XML
SIP Parameter	INVITE:	6 LII C V	ALLIE4 (DIVIT	A LILO MALLIES (DIVIT)
values:	PSTN XML HighLayerCompatibilities 180 Ringing;	ty: HLC_V	ALUE1 (PIXII)), HLC_VALUE2 (PIXII)
	PSTN XML ProgressIndicator: inte	erworking l	nas occurred a	nd has resulted in a
	telecommunication service change		ias occurred a	na nas resulted in a
	PSTN XML HighLayerCapability:		JE2 (PIXIT)	
ISUP Parameter	CPG; event information paramete	r event in	dicator: Alertir	
values:	progress indicator: interworking ha	as occurre	d and has resu	lited in a telecommunication
	service change (#5)			
	HLC: HLC_VALUE2 (PIXIT)		· -	10115
Comments:	SIP	St	JT	ISUP
	INVITE →		→	IAM
			~	ACM (no indication)
	180 Ringing ←		←	CPG (Alerting)
		Ringin	=	2. 2 (, worming)
	200 OK INVITE ←		←	ANM
	ACK →			
		Conve	rsation	
	BYE →		→	REL
	200 OK BYE ←		+	RLC

TP105013	SIP reference: RFC 320	61 [6]		ISUP reference: 27 [1], clause 7.2.3.2.1.4A
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).			
SIP selection	PICS 4/18			·
criteria:				
ISUP selection				
criteria:				
Test purpose:	CPG Progress received, sending and HLC in the 183			
SIP Parameter	the progress indication telecommunication serv HighLayerCapability .	arameter ever Parameter (A ress indicator a service change Progress mess "interworking herice change" (#	nt indicator is TP) including a #5: "interworkinge": sage with a PST has occurred are #5) and with the	set to "Progress" and High Layer Compatibility Ing has occurred and has IN XML ProgressIndicator with Ind has resulted in a PSTN XML
	INVITE: PSTN XML HighLayerCapability : HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT) 183 Session Progress; PSTN XML ProgressIndicator: interworking has occurred and has			
values:	resulted in a telecommunication service change (#5) PSTN XML HighLayerCapability: HLC_VALUE2 (PIXIT)			
ISUP Parameter	CPG; event information paran			ess
values:	ATP: progress indicator: interworking has occurred and has resulted in a			
	telecommunication service change (#5)High Layer Capability: HLC_VALUE2 (PIXIT)			
Comments:	SIP	_	JT	ISUP
	INVITE	→	→	IAM
			+	ACM (no indication)
	183 Session Progress	← Ringin	eg tone	CPG (Progress)
	200 OK INVITE	←	←	ANM
	ACK	→		
		Conve	rsation	
	BYE	→	→	REL
	200 OK BYE	←	+	RLC

TP105014	SIP reference: RFC	3261 [6]	ES 283	ISUP reference: 027 [1], clause 7.2.3.2.1.4A
TSS reference:	SIP-ISUP/Basic call/ Receipt	of the Call pro		
SIP selection criteria:	PICS 4/18			
ISUP selection criteria:				
SIP Parameter values: ISUP Parameter values:	CPG Progress containing ATP PI, mapping into PSTN XML ProgressIndicator in the sent 183 Ensure that the SUT having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "Progress", the ATP contains the progress indicator set to PI-VALUE, the BCI ISUP indicator parameter set to ISUP used all the way and the BCI ISDN access indicator set to ISDN: • sends a 183 Session Progress message with the PSTN XML ProgressIndicator information element set to PI_ID. 183 Session Progress; PSTN XML ProgressIndicator: PI_ID and "Terminating access ISDN"(#7) ACM; BCI ISUP indicator: ISUP used all the way BCI ISDN access indicator: non ISDN CPG; event information parameter event indicator: Progress BCI ISUP indicator: ISUP used all the way			
	BCI ISDN access indicator: ATP Progress Indicator value	_		
Comments:	SIP INVITE	→	SUT → ←	ACM (no indication)
	183 Session Progress	← Ring	ing tone	CPG (Progress)
	200 OK INVITE ACK	← →	←	ANM
	BYE	Con¹	ersation	REL
	200 OK BYE	-		• •==

Values and additional selection criteria for test purpose TP105014		
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)	
VA_02	PI_VALUE = Destination address is non-ISDN (#2)	

6.2.1.6 Receipt of the Answer message (ANM)

TP106001	SIP reference: R	FC 3261 [6]	50 000	ISUP reference:
				027 [1], clause 7.2.3.1.5 0 356-1 [28] clause 2.1.7
TSS reference:	SIP-ISUP/Basic call/ Red	ceipt of the Answer m		
SIP selection			errege ()	-
criteria:				
ISUP selection				
criteria:				
Test purpose:	ANM received, a 200 Ok	(INVITE is sent		
	Facility that the OUT has		4 0 -	U. d. a. a.t atat a dia ata a a at ta
	"subscriber free", on rece			lled party status indicater set to
	Subscriber free , off rece	elpt of all Alvivi filessa	ige.	
	sends a 200 OK	INVITE to the UAC.		
SIP Parameter				
values:				
ISUP Parameter				
values:				
Comments:	SIP	SUT		ISUP
	INVITE	→	→	IAM
	180 Ringing	←	+	ACM (free)
		Ringir	ng tone	
	200 OK INVITE	←	+	ANM
	ACK	→		
		Conve	rsation	
	BYE	→	→	REL
	200 OK BYE			RLC

TP106002	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.5	
		ETS 300 356-1 [28] clause 2.1.7	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer	message (ANM).	
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:			
Test purpose:	ANM received, mapping of PI contained in t	he ATP into the 200 OK PSTN XML PI	
SIP Parameter values:	Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing a progress indicator set to PI_VALUE in the ATP: • sends a 200 OK included the PSTN XML ProgressIndicator set to PI_VALUE. 200 OK; PSTN XML ProgressIndicator: PI_VALUE (PIXIT)		
ISUP Parameter values:	ANM; ATP progress indicator: PI_VALUE	(PIXIT)	
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM (free)	
	Rin	ging tone	
	200 OK INVITE ←	← ANM	
	ACK →		
	Con	versation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP106003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28] clause 2.1.7	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer m	essage (ANM).	
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:	PICS 4/19		
Test purpose:	Fallback procedure: ANM received no BC in an ATP, mapping of TMU parameter into the PSTM XML PI sent in the 200 OK INVITE Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the Transmission Medium Used set to TMU_VALUE and the ATP without Bearer Capability (BC): • sends a 200 OK message with the PSTN XML Bearer Capability encoded ISDN_BC_VALUE.		
SIP Parameter	INVITE:		
values:	PSTN XML first Bearer Capability: SETUP_		
	PSTN XML second Bearer Capability: SETU 200 OK; Bearer capability: ISDN_BC_VALUE	JP_BC2	
ISUP Parameter	ANM; TMU: TMU_VALUE		
values:	ATP: no BC		
Comments:		UT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM (free)	
		ng tone	
	200 OK INVITE	← ANM	
	ACK →		
	Conve	ersation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

	Values for test purposes TP106003				
Test purposes	ANM parameter values	200 OK parameter values	SETUP parameter values		
VA_01	TMU_VALUE: speech	ISDN_BC_VALUE: speech	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements		
VA_02	TMU_VALUE: 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements		

TP106004	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28] clause 2.1.7	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).		
SIP selection	PICS 4/18	,	
criteria:			
ISUP selection criteria:	PICS 4/19		
Test purpose:	Fallback procedure: ANM received BC in an ATP, mapping of TMU parameter and BC into the PSTM XML PI sent in the 200 OK INVITE		
	Ensure that the SUT, having received the AC containing the Transmission Medium Used ATP_VALUE and containing the progress in has resulted in a telecommunication service of	set to TMU_VALUE and the ATP set to dicator set to "interworking has occurred and	
		STN XML BearerCapability encoded	
		L ProgressIndicator set to "interworking has	
SIP Parameter	INVITE:		
values:	PSTN XML first BearerCapability: SETUP_		
	PSTN XML second Bearer Capability: SETUP_BC2		
	200 OK; PSTN XML BearerCapability: ISDN_BC_VALUE		
	ProgressIndication: interworking has occurred and has resulted in a telecommunication		
	service change(#5)		
ISUP Parameter	ANM; TMU : TMU_VALUE		
values:	ATP: ATP_VALUE		
	Progress indication: interworking has occur	red and has resulted in a telecommunication	
_	service change(#5)		
Comments:		SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM (free)	
	•	ng tone	
	200 OK INVITE ← ACK →	← ANM	
	-	ersation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	
	LOO ON DIL	TALO .	

	Values for test purposes TP106004					
Test purposes	ACM parameter values	200 OK parameter values	SETUP parameter values			
VA_01	TMU_VALUE: speech ATP_VALUE: speech	ISDN_BC_VALUE: speech and ProgressIndicator: (#5)	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements			
VA_02	TMU_VALUE: speech ATP_VALUE: 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio and ProgressIndicator : (#5)	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements			
VA_03	TMU_VALUE: 3,1 kHz audio ATP_VALUE: speech	ISDN_BC_VALUE: speech and ProgressIndicator: (#5)	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements			
VA_04	TMU_VALUE: 3,1 kHz audio ATP_VALUE: 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio and ProgressIndicator : (#5)	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements			

TP106005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5		
		ETS 300 356-1 [28], clause 2.1.7		
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer m	essage (ANM).		
SIP selection	PICS 4/18			
criteria:				
ISUP selection criteria:	PICS 4/19			
Test purpose:	Fallback procedure: ANM received contains an ATP with BC unrestricted digital information with tones/announcement, mapping into the PSTN XML PI in the sent 200 OK INVITE			
	Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the ATP including the Bearer Capability set to "unrestricted digital information with tones/announcement" and without TMU parameter: • sends a 200 OK message with the PSTN XML Bearer Capability set to "unrestricted digital information with tones/announcement".			
SIP Parameter	200 OK; PSTN XML BearerCapabilty: unrestricted digital information with			
values:	tones/announcement			
ISUP Parameter	ANM; ATP BC: unrestricted digital information with tones/announcement			
values:	no TMU	LIT. IOLID		
Comments:		SUT ISUP		
	INVITE -	→ IAM		
	180 Ringing ←	← ACM (free)		
		ng tone		
	200 OK INVITE ← ACK →	← ANM		
	-	a wa ati a w		
	BYE →	ersation REL		
	200 OK BYE ←	← RLC		

TP106006	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5		
		ETS 300 356-1 [28], clause 2.1.7		
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer	message (ANM).		
SIP selection criteria:	PICS 4/18			
ISUP selection criteria:				
Test purpose:	ANM received contains an ATP parameter, mapping of HLC into the PSTN HLC in the 200 OK INVITE Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the HLC parameter in the ATP set to HLC_VALUE: • sends a 200 OK message PSTN XML HighLayerCompatibility information element set to HLC_VALUE.			
SIP Parameter	200 OK;			
values:	PSTN XML HighLayerCompatibility: HLC_VALUE (PIXIT)			
ISUP Parameter	ANM; ATP HLC: HLC_VALUE			
values:	,			
Comments:	SIP	SUT ISUP		
	INVITE ->	→ IAM		
	180 Ringing ←	← ACM (free)		
	5 5	ging tone		
	200 OK INVITE ←	← ANM		
	ACK →			
	Cor	versation		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

TP106007	SIP reference: RFC 3261 [6]	ES 283 (ETS 300	SUP reference: 027 [1], clause 7.2.3.1.5 356-1 [28] clause 2.1.7	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer r	nessage (ANM).		
SIP selection criteria:	PICS 4/18			
ISUP selection criteria:	PICS 4/19			
SIP Parameter values:	Fallback procedure: ANM received contains an ATP with HLC and PI #5, mapping into the PSTN XML PI in the sent 200 OK INVITE Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the HLC parameter in the ATP with an HLC set to HLC_VALUE and the progress indicator set to "interworking has occurred and has resulted in a telecommunication service change" (#5): • sends a 200 OK message with the PSTN XML HighLayerCompatibility information element set to HLC_VALUE and the progress indicator set to "interworking has occurred and has resulted in a telecommunication service change" (#5). INVITE: PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT) 200 OK; PSTN XML HighLayerCompatibility: HLC_VALUE2 (PIXIT)			
	PSTN XML ProgressIndicator: interworking			
	telecommunication service change (#5)			
ISUP Parameter	ANM; ATP HLC: HLC_VALUE2			
values:	progress indicator : interworking has occurred and has resulted in a telecommunication service change (#5)			
Comments:		SUT	ISUP	
	INVITE →	→	IAM	
	180 Ringing ←	←	ACM (free)	
		ng tone		
	200 OK INVITE ←	←	ANM	
	ACK →			
		ersation	DEI	
	BYE -	→	REL	
	200 OK BYE ←	+	RLC	

TP106008	SIP reference: RFC 3261	[6]		ISUP reference: 027 [1], clause 7.2.3.1.5 0 356-1 [28] clause 2.1.7
TSS reference:	SIP-ISUP/Basic call/ Receipt of the	e Answer m	essage (ANM).	
SIP selection	PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	ANM received contains an ATP conveying the LLC, mapping into the PSTN XML LLC in the sent 200 OK INVITE Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the LLC parameter in the ATP set to LLC_VALUE: • sends a 200 OK message with a PSTN XML LowLayerCompatibility information element set to LLC_VALUE.			
SIP Parameter values:	200 OK INVITE: PSTN XML Low	LayerComp	eatibility: LLC_	VALUE (PIXIT)
ISUP Parameter values:	ANM; ATP LLC: LLC_VALUE			
Comments:	SIP	S	UT	ISUP
	INVITE -	•	→	IAM
	180 Ringing	•	←	ACM (free)
		Ringi	ng tone	
	200 OK INVITE	•	←	ANM
	ACK -	•		
		Conve	ersation	
	BYE -	•	→	REL
	200 OK BYE ★		+	RLC

6.2.1.7 Receipt of the Connect message (CON)

TP107001	SIP re	ference: RFC 3261 [6]	ES 28	ISUP reference: 3 027 [1], clause 7.2.3.1.5	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).				
SIP selection criteria:	PICS 4/1 AND PICS 4/4 AND PICS 4/5				
ISUP selection criteria:					
Test purpose:	CON received, 200 OK INVITE is sent after Preconidions are met				
	SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON message:				
	sends a 200 OK INVITE to the UAC. The bearer path shall be connected in both directions when both of the following conditions are satisfied:				
	 the I-MGCF determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). 				
	In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient				
SIP Parameter values:	preconditions have been met for the session to proceed. INVITE: Require: precondition SDP a=curr:qos local none				
	183: Require: 100rel SDP				
	UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv				
	200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv				
ISUP Parameter values:					
Comments:	SIP		SUT	ISUP	
	INVITE 183 Session F PRACK	Progress	-2	IAM	
	200 OK PRAC UPDATE	→	-3	COT(successful)	
	200 OK UPDA 200 OK INVIT		•	- CON	
	ACK	→			
	DVE		nversation	A DEI	
	BYE 200 OK BYE	→	- 2	REL FRLC	
L	LOO ON DIL			i illo	

TP107002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5				
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).					
SIP selection	NOT PICS 4/1 AND PICS 4/4 AND PICS 4/5					
criteria:						
ISUP selection criteria:						
Test purpose:	CON received, 200 OK INVITE is sent after Preconidions are met					
	SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON message:					
	sends a 200 OK INVITE to the UAC. The bearer path shall be connected in both directions when both of the following conditions are satisfied:					
	 the I-MGCF determines (using the pro sufficient preconditions have been sat establishment to proceed (if applicable 	isfied on the SIP side for session				
	Outgoing bearer set-up procedure and the Co bearer path shall be connected in both directic and the I-IWU determines (through the proced preconditions have been met for the session to	addition, if BICC is performing the "Per-call bearer set-up in the forward direction" agoing bearer set-up procedure and the Connect Type is "notification not required", the arer path shall be connected in both directions when the Bearer Set-up request is sent at the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient conditions have been met for the session to proceed.				
SIP Parameter	INVITE: Require: precondition					
values:	SDP a=curr:qos local none					
	a=curr:qos remote none a=des:qos mandatory local send	drecv				
	a=des.qos mandatory local sendrecv a=des:qos none remote sendrecv					
	183: Require: 100rel					
	SDP a=curr:qos local none					
	a=curr:qos remote none					
	a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv					
	a=des.qos mandatory remote sendrecv a=conf:qos remote sendrecv					
	UPDATE:					
	SDP a=curr:qos local sendrecv					
	a=curr:qos remote none					
	a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv					
	200 OK UPDATE					
	SDP a=curr:qos local sendrecv					
	a=curr:qos remote sendrecv					
	a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv					
ISUP Parameter	a -accided mandatory remote of					
values:	loup	10:15				
Comments:	SIP SINVITE →	UT ISUP				
	183 Session Progress					
	PRACK →					
	200 OK PRACK ←					
	UPDATE 200 OK UPDATE ←	→ IAM				
	200 OK UPDATE 200 OK INVITE ←	← CON				
	ACK →					
	Conversation					
	BYE -	→ REL				
	200 OK BYE ←	← RLC				

TP107003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5			
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	CON received, 200 OK INVITE is sent				
	SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON message: sends a 200 OK INVITE to the UAC. The bearer path shall be connected in both directions. In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "notification not required", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient				
SIP Parameter	preconditions have been met for the session to proceed. INVITE: SDP offer				
values:	200 OK INVITE: SDP answer				
ISUP Parameter values:					
Comments:	SIP	SUT ISUP			
	INVITE ->	→ IAM			
	200 OK INVITE ←	← CON			
	ACK →				
		ersation			
	BYE →	→ REL			
	200 OK BYE ←	← RLC			

TP107004	SIP reference: RFC 326	1 [6]		ISUP reference: 027 [1], clause 7.2.3.1.5 0 356-1 [28], clause 2.1.7
TSS reference:	SIP-ISUP/Basic call/ Receipt of	the Answer m		
SIP selection criteria:	PICS 4/18			
ISUP selection criteria:				
Test purpose:	CON received contains a PI conveyed in an ATP, mapped into the PSTN XML PI sent in the 200 OK INVITE			
	Ensure that the SUT, on receipt of an CON message containing a progress indicator set to PI_VALUE in the ATP: • sends a 200 OK included the PSTN XML ProgressIndicator set to PI_VALUE.			
SIP Parameter values:	200 OK; PSTN XML ProgressIndicator: PI_VALUE (PIXIT)			
ISUP Parameter values:	ANM; ATP progress indicator: PI_VALUE (PIXIT)			
Comments:	SIP	S	UT	ISUP
	INVITE	→	→	IAM
		←	←	CON
	ACK	→		
	Conversation			
	-·-	→	→	REL
	200 OK BYE	-	-	RLC

TP107005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28], clause 2.1.7	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer m	essage (CON).	
SIP selection	PICS 4/18		
criteria:			
ISUP selection criteria:	PICS 4/19		
Test purpose: SIP Parameter values:	CON received contains the TMU parameter, mapping into the PSTN XML BC sent in the 200 OK INVITE Ensure that the SUT, on receipt of an CON message containing the Transmission Medium Used set to TMU_VALUE and the ATP without Bearer Capability (BC): • sends a 200 OK message with the PSTN XML BearerCapability encoded ISDN_BC_VALUE. INVITE: PSTN XML first Bearer Capability: SETUP_BC1 PSTN XML second Bearer Capability: SETUP_BC2		
	200 OK; Bearer capability: ISDN_BC_VALUE		
ISUP Parameter	CON; TMU: TMU_VALUE		
values:	ATP: no BC		
Comments:		UT ISUP	
	INVITE ->	→ IAM	
	200 OK INVITE ←	← CON	
	ACK →		
	Conversation		
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

Values for test purposes TP107005			
Test purposes	CON parameter values	200 OK parameter values	SETUP parameter values
A_01	TMU_VALUE: speech	ISDN_BC_VALUE: speech	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements
/A_02	TMU_VALUE: 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements

TP107006	SIP reference: RFC 3261 [6]	ISUP reference:		
		ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28], clause 2.1.7		
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (CON).			
SIP selection	PICS 4/18			
criteria:				
ISUP selection	PICS 4/19			
criteria:				
Test purpose:	CON received contains the TMU parameter at			
	mapping into the PSTN XML PI sent in the 20	O OK INVITE		
	Ensure that the SUT, on receipt of an CON m	essage containing the Transmission		
	Medium Used set to TMU_VALUE and the A			
	progress indicator set to "interworking has o	occurred and has resulted in a		
	telecommunication service change" (#5):			
		sends a 200 OK message with the PSTN XML BearerCapability encoded		
	ISDN_BC_VALUE and the PSTN XML ProgressIndicator set to "interworking has			
SIP Parameter	occurred and has resulted in a telecommunication service change" (#5).			
values:	PSTN XML first BearerCapability: SETUP_BC1			
	second Bearer Capability: SETUP_BC2			
	200 OK;			
	PSTN XML BearerCapability: ISDN_BC_VALUE			
	ProgressIndication: interworking has occurred and has resulted in a telecommunication			
10115 5	service change(#5)			
ISUP Parameter values:	CON; TMU: TMU_VALUE			
values.	ATP: BC ATP_VALUE Progress indication: interworking has occurred and has resulted in a telecommunication			
	service change(#5)			
Comments:	0 \ /	UT ISUP		
	INVITE →	→ IAM		
	200 OK INVITE ←	← CON		
	ACK →			
	Conve	rsation		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

	Values for test purposes TP107006			
Test purposes	ACM parameter values	200 OK parameter values	SETUP parameter values	
VA_01	TMU_VALUE: speech ATP_VALUE: BC speech	ISDN_BC_VALUE: speech and ProgressIndicator: (#5)	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements	
VA_02	TMU_VALUE: speech ATP_VALUE: BC 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio and ProgressIndicator : (#5)	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements	
VA_03	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC speech	ISDN_BC_VALUE: speech and ProgressIndicator: (#5)	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements	
VA_04	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio and ProgressIndicator : (#5)	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements	

TP107007	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5	
		ETS 300 356-1 [28] clause 2.1.7	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer r	message (CON).	
SIP selection	PICS 4/18		
criteria:			
ISUP selection criteria:	PICS 4/19		
Test purpose:	CON received contains the ATP parameter conveying the BC unrestricted digital information with tones/announcement, mapping into the PSTN XML BC sent in the 200 OK INVITE		
	Ensure that the SUT, on receipt of an CON message containing the ATP including the Bearer Capability set to "unrestricted digital information with tones/announcement" and without TMU parameter: • sends a 200 OK message with the PSTN XML Bearer Capability set to "unrestricted digital information with tones/announcement".		
SIP Parameter	200 OK; PSTN XML BearerCapabilty: unrestricted digital information with		
values:	tones/announcement		
ISUP Parameter	CON; ATP BC: unrestricted digital information with tones/announcement		
values:	no TMU		
Comments:			
	INVITE -	→ IAM	
	200 OK INVITE ←	← CON	
	ACK →		
		versation	
	BYE -	→ REL	
	200 OK BYE ←	← RLC	

TP107008	SIP reference: RFC 326	1 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.	.5
			ETS 300 356-1 [28], clause 2.1	.7
TSS reference:	SIP-ISUP/Basic call/ Receipt of	the Answer m	essage (CON).	
SIP selection	PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:			onveying the HLC, mapping into the PS	STN
	XML HLC sent in the 200 OK IN	VITE		
	E II III OUT	(00N		,
		of an CON m	essage containing the HLC parameter i	n the
	ATP set to HLC_VALUE:			
	and a 200 OK massage BCTN VMI High aver Commedibility in farmenting			
	 sends a 200 OK message PSTN XML HighLayerCompatibility information element set to HLC_VALUE. 			
SIP Parameter	200 OK:			
values:	PSTN XML HighLayerCompatibility: HLC_VALUE (PIXIT)			
ISUP Parameter	CON; ATP HLC: HLC_VALUE			
values:				
Comments:	SIP	S	UT ISUP	
	INVITE	→	→ IAM	
	200 OK INVITE	←	← CON	
	ACK	→		
	Conversation			
	BYE	→	→ REL	
	200 OK BYE	-	← RLC	

TP107009	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28], clause 2.1.7	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer m	nessage (CON).	
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:	PICS 4/19		
Test purpose: SIP Parameter values:	CON received contains the ATP parameter conveying the HLC and PI #5, mapping into the PSTN XML HLC sent in the 200 OK INVITE Ensure that the SUT, on receipt of an CON message containing the HLC parameter in the ATP with an High Layer Compatibility set to HLC_VALUE and the Progress.indicator set to "interworking has occurred and has resulted in a telecommunication service change" (#5): • sends a 200 OK message with the PSTN XML HighLayerCompatibility information element set to HLC_VALUE and the progress indicator set to "interworking has occurred and has resulted in a telecommunication service change" (#5). INVITE: PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT) 200 OK; PSTN XML HighLayerCompatibility: HLC_VALUE2 (PIXIT)		
	progress indicator: interworking has occurred and has resulted in a telecommunication		
	service change (#5)		
ISUP Parameter	CON; ATP HLC: HLC_VALUE2		
values:	progress indicator: interworking has occurred and has resulted in a telecommunication		
0	service change (#5)		
Comments:		SUT ISUP	
	INVITE -	→ IAM	
	200 OK INVITE	← CON	
	ACK →	orgation	
		ersation → REL	
	BYE 200 OK BYE	→ REL ← RLC	
	ZUU UN DIE T	▼ KLU	

TP107010	SIP reference: RFC	3261 [6]	ISUP reference:
			ES 283 027 [1], clause 7.2.3.1.5
TSS reference:	ETS 300 356-1 [28] clause 2.1.7 SIP-ISUP/Basic call/ Receipt of the Answer message (CON).		
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:			
Test purpose:	CON received contains an ATP conveying the LLC, mapping into the PSTN XML sent in the 200 OK INVITE Ensure that the SUT, having received the ACM message, on receipt of an CON message containing the LLC parameter in the ATP set to LLC_VALUE: • sends a 200 OK message wit a PSTN XML LowLayerCompatibility information element set to LLC_VALUE.		
SIP Parameter values:	200 OK; PSTN XML LowLay	erCompatibility	y: LLC_VALUE (PIXIT)
ISUP Parameter values:	CON; ATP LLC: LLC_VALUE		
Comments:	SIP INVITE 200 OK INVITE ACK BYE 200 OK BYE	→ ← → Con	SUT ISUP → IAM ← CON nversation → REL ← RLC

6.2.1.8 Receipt of the REL message

TP108001	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8	
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release r	message (REL)/	
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	REL received after IAM was sent. Mapping int	to final response containing a Reason header	
OID D	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, on receipt of an ISUP REL, where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • The ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field.		
SIP Parameter values:	cause value: CV_SIP (PIXIT)		
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)		
Comments:	SIP SU	IT ISUP	
	INVITE →	→ IAM	
	SIP_FAILURE_VA ←	← REL	
	ACK →	→ RLC	

Table 4

	Values for test purposes TP108001			
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISUP		
VA_1	404 Not Found Cause Value No. 1	Cause Value No. 1 ("unallocated (unassigned) number")		
VA_2	500 Server Internal Error Cause Value No. 2	Cause Value No. 2 ("no route to network")		
VA_3	500 Server Internal Error Cause Value No. 3	Cause Value No. 3 ("no route to destination")		
VA_4	500 Server Internal Error Cause Value No. 4	Cause Value No. 4 ("Send special information tone")		
VA_5	404 Not Found Cause Value No. 5	Cause Value No. 5 ("Misdialled trunk prefix")		
VA_6	500 Server Internal Error Cause Value No. 8	Cause Value No. 8 ("Preemption")		
VA_7	500 Server Internal Error Cause Value No. 9	Cause Value No. 9 ("Preemption-circuit reserved for reuse")		
VA_8	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")		
VA_9	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("no user responding")		
VA_10	480 Temporarily unavailable Cause Value No. 19	Cause Value No. 19 ("no answer from the user")		
VA_11	480 Temporarily unavailable Cause Value No. 20	Cause Value No. 20 ("subscriber absent")		
VA_12	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")		
VA_13	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")		
VA_14	480 Temporarily unavailable Cause Value No. 25	Cause Value No. 25 ("Exchange routing error")		
VA_15	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")		
VA_16	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete"))		
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")		
VA_18	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)		
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable Cause Value No. 34	Cause Value in the Class 010 (resource unavailable, Cause Value No. 34)		
VA_20	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)		
VA_21	500 Server Internal Error Cause Value No. 50	Cause Value No. 50 ("requested facility not subscribed")		
VA_22	500 Server Internal Error Cause Value No. 55	Cause Value No. 55 ("incoming calls barred within CUG")		
VA_23	500 Server Internal Error Cause Value No. 57	Cause Value No. 57 ("bearer capability not authorized")		
VA_24	500 Server Internal Error Cause Value No. 58	Cause Value No. 58 ("bearer capability not presently")		
VA_25	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)		
VA_26	500 Server Internal Error Cause Value No. 65 - 79	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 - 79) (79 is class default)		
VA_27	500 Server Internal Error Cause Value No. 87	Cause Value No. 87 ("user not member of CUG")		

	Values for test purposes TP108001		
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISUP	
VA_28	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")	
VA_29	500 Server Internal Error Cause Value No. 90	Cause Value No. 90 ("Non-existent CUG")	
VA_30	404 Not Found Cause Value No. 91	Cause Value No. 91 ("invalid transit network selection")	
VA_31	500 Server Internal Error Cause Value No. 95	Cause Value No. 95 ("invalid message") (Class default)	
VA_32	500 Server Internal Error Cause Value No. 97	Cause Value No. 97 ("Message type non-existent or not implemented")	
VA_33	500 Server Internal Error Cause Value No. 99	Cause Value No. 99 ("information element/parameter non- existent or not implemented")	
VA_34	480 Temporarily unavailable Cause Value No. 102	Cause Value No. 102 ("recovery on timer expiry")	
VA_35	500 Server Internal Error Cause Value No. 103	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")	
VA_36	500 Server Internal Error Cause Value No. 110	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")	
VA_37	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)	
VA_38	480 Temporarily unavailable Cause Value No. 127	Cause Value No. 127 ("interworking unspecified") (Class default)	

TP108002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8	
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/		
SIP selection criteria:	PICS 4/15	noodago (NEE)	
ISUP selection criteria:			
Test purpose:	REL after ACM received, mapping in a final response Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "no indication", on receipt of an ISUP REL, where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason		
	header field.		
SIP Parameter values:	cause value: CV_SIP (PIXIT)		
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)		
Comments:	SIP SU INVITE → SIP_FAILURE_VA ← ACK →	ISUP → IAM ← ACM (no indication) ← REL → RLC	

Table 5

	Values for test purpose TP108002				
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISUP,			
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")			
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")			
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")			
VA_4	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")			
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")			
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")			
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)			
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)			
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)			
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")			
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)			

TP108003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8		
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release i			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	REL received in the early dialogue (ACM free) mapping in a final response Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "subscriber free", having sent a 180 Ringing message on receipt of an ISUP REL, where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason			
SIP Parameter	header field. Cause value: CV_SIP (PIXIT)			
values:	(1741)			
ISUP Parameter values:	REL; Cause value: CV_ISUP (PIXIT)			
Comments:	SIP	JT ISUP		
	INVITE ->	→ IAM		
	180 Ringing ←	← ACM (free)		
	SIP_FAILURE_VA	← REL		
	ACK →	→ RLC		

TP108004	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8		
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/			
SIP selection	NOT PICS 4/15	necouge (NEE)		
criteria:				
ISUP selection				
criteria:				
Test purpose:	REL received in the early dialogue (CPG Aler	ting) mapping in a final response		
	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "no indication", having received a CPG message where the event information parameter event indicator is set to "Alerting", a 180 Ringing message is sent, on receipt of an where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field.			
SIP Parameter	Cause value: CV_SIP (PIXIT)			
values: ISUP Parameter	REL; cause value: CV_ISUP (PIXIT)			
values:	INCL, Cause value. CV_ISUF (FIXIT)			
Comments:	SIP SU	JT ISUP		
	INVITE ->	→ IAM		
		 ACM (no indication) 		
	180 Ringing ←	← CPG (Alerting)		
	SIP_FAILURE_VA	← REL		
	ACK →	→ RLC		

Table 6

	Values for test purposes TP108003 and TP108004				
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISUP,			
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")			
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")			
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")			
VA_4	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")			
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")			
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")			
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)			
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)			
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)			
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")			
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)			

TP108005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8		
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	REL received in the confirmed state (ANM rec	ceived)		
SIP Parameter	 Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message, having received an ANM', a 200 OK message is sent, on receipt of an ISUP REL where the cause value defined as CV_ISUP: the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; the SUT shall send BYE message; the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field in the BYE. 			
values:	Cause value: CV_SIP (PIXIT)			
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)			
Comments:	SIP SU INVITE → 180 Ringing ← 200 OK INVITE ← ACK → Convei	→ IAM ← ACM ← ANM		

TP108006	SIP reference: RFC 3261 [6]	ISUP reference:			
11 100000		ES 283 027 [1], clause 7.2.3.1.8			
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/				
SIP selection	·	· · · · · · · · · · · · · · · · · · ·			
criteria:					
ISUP selection					
criteria:					
Test purpose:	REL received in the confirmed state (CON received)				
	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a IAM message, having received a CON message, a 200 OK message is sent, on receipt of an where the cause value defined as CV_ISUP:				
	 the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; the SUT shall send BYE message. the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field. 				
SIP Parameter values:	Cause value: CV_SIP (PIXIT)				
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)				
Comments:	SIP SUT ISUP				
	INVITE →	→ IAM			
	200 OK INVITE ←	← CON			
	ACK →				
	Conversation				
	BYE ← REL				
	200 OK BYE →	→ RLC			

Table 7

	Values for test purposes TP108005 and TP108006				
←SIP Message SIP_FAILURE_VA CV_SIP		← REL Cause Indicators parameter CV_ISUP,			
VA_1	BYE Cause Value No. 16	Cause Value No. 16			
VA_2	BYE Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)			
VA_3	BYE Cause Value No. 38	Cause Value No. 38 ("Network out of order")			
VA_4	BYE Cause Value No. 41	Cause Value No. 41 ("Temporary failure ")			
VA_5	BYE Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)			

6.2.1.9 Autonomous release at I-MGCF

TP109001	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.10			
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF				
SIP selection criteria:					
ISUP selection criteria:	PICS 3/3 AND NOT PICS 3/4				
Test purpose:	Overlap not supported, 484 is sent if insufficient digits received in the INVITE Ensure that the SUT on receipt of insufficient digits received in an INVITE messages: • sends an 484 Address Incomplete message.				
SIP Parameter values:					
ISUP Parameter values:					
Comments:	SIP INVITE → 484 Address incomplete ← ACK →	SUT ISUP			

TP109002	SIP reference: RFC 326	1 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.10		
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	480 is sent if congestion in the S	UT			
SIP Parameter values:	Ensure that the SUT in congestion on receipt of INVITE message: sends an 480 Temporarily Unavailable message.				
ISUP Parameter values:					
Comments:	SIP INVITE	SI →	SUT ISUP		
	ioo ionipolani, anaranaoio	← →			

TP109003	SIP reference: RFC 32	61 [6]		ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.8.10			
TSS reference:	SIP-ISUP/Basic call/ Autonomo	us release at I-	-MGCF		
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	500 is sent to due the compatib	ility procedure	for unknown pa	arameters	
SIP Parameter values:	Ensure that the call is released due to the BICC/ISUP compatibility procedure for unknown parameters: • sends 500 Server Internal Error.				
ISUP Parameter values:	Unknown parameter in ACM: Parameter compatibility "Release call"				
Comments:	SIP SUT ISUP				
	INVITE	→	→	IAM	
			←	ACM (???)	
	500 Server internal error	←	→	REL	
	ACK → ← RLC				

TP109004	SIP reference: RFC 3261 [6]	ISUP reference:			
		ES 283 027 [1], clause 7.2.3.1.8.10			
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	Call setup is cleared after T7 expiry				
	Ensure that the call is released due to expiry of T7 within the BICC/ISUP procedures:				
	sends 484 Address Incomplete.				
SIP Parameter					
values:					
ISUP Parameter					
values:					
Comments:	SIP	SUT ISUP			
	INVITE →	→ IAM			
	Expiry of T7				
	484 Address incomplete ←	→ REL			
	ACK → RLC				

TP109005	SIP reference: RFC 32	261 [6]		ISUP reference:	
			ES 283 0	27 [1], clause 7.2.3.1.8.10	
TSS reference:	SIP-ISUP/Basic call/ Autonomo	ous release a	at I-MGCF		
SIP selection criteria:					
ISUP selection criteria:	PICS 4/16				
Test purpose:	Call setup is cleared after T9 e	expiry			
SIP Parameter values:	Ensure that the call is released due expiry of T9 within the BICC/ISUP procedures: sends 480 Temporarily Unavailable.				
ISUP Parameter					
values:					
Comments:	SIP SUT ISUP				
	INVITE → IAM				
	180 Ringing ← ← ACM				
	Expiry of T9				
	480 Temporarily unavailable ← → REL				
	ACK	→	+	RLC	

TP109006	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.8.10	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-	-MGCF	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	500 is sent to due the compatibility procedure	for unknown messages	
	Ensure that the call is released due to the BICC/ISUP compatibility procedure for unknown		
	messages:		
	 sends 500 Server Internal Error. 		
SIP Parameter			
values:			
ISUP Parameter	XXX: Unknown message: message compatibi	lity "Release call"	
values:			
Comments:	SIP	UT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		← XXX	
	500 Server internal error ←	→ REL	
		· -—	
	ACK →	← RLC	

6.2.1.10 Receipt of the Release message BYE / CANCEL

TP110001	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clause 7.2.3.1.6
TSS reference:	SIP-ISUP/Basic call/ Receipt of the BYE mes	sage
SIP selection		
criteria:		
ISUP selection		
criteria:		
Test purpose:	BYE with Reason header received, sending of	f REL
	Ensure that the SUT on receipt of SIP BYE, the side:	ne SUT shall send an ISUP REL to the ISUP
	included in the BYE message is ma ISUP REL message with the location	with ITU-T Rec Q.850 [5] Cause Value is pped to the ISUP Cause Value field in the network beyond interworking point".
SIP Parameter values:	Protocol-cause: CV_Reason Header (PIXIT)	
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT)	
Comments:	SIP	UT ISUP
	INVITE ->	→ IAM
	180 Ringing ←	← ACM
	200 OK INVITE ←	← ANM
	ACK →	7 AVVI
	BYE →	→ REL
	200 OK BYE ←	← RLC

TP110002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.6	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CANCEL r	·	
SIP selection	Cit 1001 / Baolo calli 1 (coolpt ci tilo ci tilo ci	necoage	
criteria:			
ISUP selection			
criteria:			
Test purpose:	CANCEL with Reason header received, sendi	ng of REL	
	5 4 44 907		
	ISUP side.	EL, the I-MGCF shall send an ISUP REL to the	
	Ensure that the Reason header field with ITU-T Rec Q.850 [5] Cause Value is		
	included in the CANCEL message is mapped to the ISUP Cause Value field in the		
	ISUP REL message with the location	"network beyond interworking point".	
SIP Parameter			
values:			
ISUP Parameter	REL: cause value: CV_ISUP (PIXIT)		
values:	location: "network beyond interworking point	II	
Comments:	SIP	UT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	CANCEL →	→ REL	
	200 OK CANCEL	← RLC	
		₹ KLU	
	487 Request Terminated ← ACK →		

TP110003	SIP reference: RFC 3261	[6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.6
<i>(</i>		->	:
TSS reference:	SIP-ISUP/Basic call/ Receipt of th	e BYE mess	sage
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	BYE without Reason header recei	ved, sendin	g of REL
	Ensure that the SUT on receipt of	SIP BYE wi	thout Reason header, the SUT shall send an
	ISUP REL to the ISUP side.		·
	Ensure that the the coding of the ISUP Cause Value is # 16 with the location "network		
	beyond interworking point" if no reason header is containd in the SIP message.		
SIP Parameter			
values:			
ISUP Parameter	REL: cause value: #16		
values:			
Comments:	SIP	S	UT ISUP
	INVITE -	→	→ IAM
	180 Ringing	-	← ACM
		F	← ANM
		→	2 /
	AON	•	
	BYE -	>	→ REL
		=	
	200 OK BYE	<u> </u>	← RLC

TP110004	SIP reference: RFC 3261 [6]		ISUP reference: 027 [1], clause 7.2.3.1.6
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CANCEL		027 [1], 010000 7.2.0.110
SIP selection criteria:	'		
ISUP selection criteria:			
Test purpose:	CANCEL without Reason header received, se	ending of REL	
	Ensure that the SUT on receipt of SIP CANCI an ISUP REL to the ISUP side. Ensure that the the coding of the ISUP Cause beyond interworking point" if no reason head	· Value is # 31 w	vith the location "network
SIP Parameter			-
values:			
ISUP Parameter	REL: cause value: CV_ISUP (PIXIT)		
values:	location: LOC_ISUP (PIXIT)		
Comments:	SIP	UT	ISUP
	INVITE →	→	IAM
	180 Ringing ←	←	ACM
	CANCEL →	→	REL
	200 OK CANCEL ←	←	RLC
	487 Request Terminated ←		
	ACK →		

6.2.1.11 Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented

TP111001	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clause 7.2.3.1.8.9
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message	
	(GRS) or Circuit group blocking message (C	GB) with the indication hardware failure oriented
SIP selection criteria:		
ISUP selection		
criteria:		
Test purpose:	RSC received in the confirmed state, a BYE	is sent
	Ensure that the SUT, when the communication is in the confirmed state, on receipt of a RSC message sends: • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.	
SIP Parameter		
values:		
ISUP Parameter		
values:		
Comments:	SIP	SUT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	200 OK INVITE ←	← ANM
	ACK →	
	Cor	versation
	BYE ←	← RSC
	200 OK BYE →	→ RLC

TP111002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9
TSS reference:		t message (RSC), Circuit group reset message (GB) with the indication hardware failure oriented
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	GRS received in the confirmed state, a BYE is sent Ensure that the SUT, when the communication is in the confirmed state, on receipt of a GRS message sends: • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK BYE Coi	SUT ISUP → IAM ← ACM ← ANM ANM Oversation ← GRS → GRA

TP111003	SIP reference: RFC 3261 [6]	ISUP reference:
TCC veference:	OID IOLID/Daria and Apparatus in a situation of the contraction of the	ES 283 027 [1], clause 7.2.3.1.8.9
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit n	
SIP selection	(GRS) or Circuit group blocking message (CG	B) with the indication hardware failure oriented
criteria:		
ISUP selection		
criteria:		
Test purpose:	CGB "Hardware failure oriented" received in the	he confirmed state, a BYE is sent
SIP Parameter	message, with the Circuit Group Supervision Mailure oriented", sends:	n is in the confirmed state, on receipt of a CGB Message Type Indicator coded as "hardware dy received an ACK for the 200 OK INVITE
values:		
ISUP Parameter	Circuit Group Supervision Message Type India	cator "hardware failure oriented"
values:	gy	
Comments:	SIP	SUT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	200 OK INVITE	← ANM
	ACK →	
	Conv	ersation
	BYE ←	← CGB
	200 OK BYE →	→ CGBA

TP111004	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.8.9	
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit	message (RSC), Circuit group reset message	
	(GRS) or Circuit group blocking message (CC	GB) with the indication hardware failure oriented	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	RSC received after 200 OK INVITE was sent	and ACK is not received	
	Ensure that the SUT, when an ANM was rece		
	200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE.		
	4 017 1 11 11 11 11 11 11 11 11 11 11 11 11		
	the SUT shall wait until it receives the ACK for the 200 OK INVITE before sending		
SIP Parameter	the BYE.		
values:			
ISUP Parameter			
values:			
Comments:	SIP	SUT ISUP	
Comments.	INVITE →	→ IAM	
		- " ""	
	180 Ringing ←	← ACM	
	200 OK INVITE ←	← ANM	
	-	← RSC	
	ACK →	→ RLC	
	BYE ←		
	200 OK BYE →		

TP111005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9
TSS reference:		circuit message (RSC), Circuit group reset message age (CGB) with the indication hardware failure oriented
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	200 OK INVITE if the SUT has not yet	ras sent and ACK is not received ras received, on receipt of a GRS message sends received an ACK for the 200 OK INVITE. receives the ACK for the 200 OK INVITE before sending
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← ACK → BYE ← 200 OK BYE →	SUT ISUP → IAM ← ACM ← ANM ← GRS → GRA

TP111006	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clause 7.2.3.1.8.9
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit n	nessage (RSC), Circuit group reset message
	(GRS) or Circuit group blocking message (CG	B) with the indication hardware failure oriented
SIP selection		
criteria:		
ISUP selection		
criteria:		
Test purpose:	CGB "Hardware failure oriented" received afte received	r 200 OK INVITE was sent and ACK is not
	Ensure that the SUT, when an ANM was received, on receipt of a RSC message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE. • The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.	
SIP Parameter values:		
ISUP Parameter	Circuit Group Supervision Message Type Indicator "hardware failure oriented"	
values:		
Comments:	SIP	UT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	200 OK INVITE	← ANM
		← CGB
	ACK →	→ CGBA
	BYE ←	
	200 OK BYE →	

TP111007	SIP reference: RFC 3261 [6]	ISUP reference:
		ES 283 027 [1], clause 7.2.3.1.8.9
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit n	
	(GRS) or Circuit group blocking message (CG	B) with the indication hardware failure oriented
SIP selection		
criteria:		
ISUP selection		
criteria:		
Test purpose:	RSC in the early dialogue received, a 500 is s	ent
		ard ISUP/BICC message relating to the call has
	already been received on receipt of a RSC message sends:	
	 a 500 Server Internal Error on the SII 	P side.
SIP Parameter		
values:		
ISUP		
Parameter		
values:		
Comments:	SIP	UT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
		← RSC
	480 Temporarily Unavailable	→ RLC
	ACK	2 1120
	,,,,,,	

TP111008	SIP reference: RFC 32	61 [6]		ISUP reference: 027 [1], clause 7.2.3.1.8.9
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection criteria:	(Cree, or one on group zreening	eseage (e-e		
ISUP selection criteria:				
Test purpose:	GRS in the early dialogue recei	ved, a 500 is s	ent	
	Ensure that the SUT, when at leadready been received on receipt a 500 Server Internal E	ot of a GRS me	essage sends:	message relating to the call has
SIP Parameter				
values:				
ISUP				
Parameter				
values:				
Comments:	SIP	5	SUT	ISUP
	INVITE	→	→	IAM
	180 Ringing	←	←	ACM
		_	(GRS
	480 Temporarily Unavailable	(→	GRA
	ACK	→		

TP111009	SIP reference: RFC 3261 [6]	ISUP reference:	
T00 (ES 283 027 [1], clause 7.2.3.1.8.9	
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit n		
	(GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented		
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	CGB "Hardware failure oriented" in the early of	lialogue received, a 500 is sent	
	Ensure that the SUT, when at least one backw		
	already been received on receipt of a CGB me		
	Message Type Indicator coded as "hardware f	ailure oriented", sends	
	a 500 Server Internal Error on the SIP side.		
SIP Parameter			
values:			
ISUP			
Parameter			
values:			
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		← CGB	
	480 Temporarily Unavailable ←	→ CGBA	
	ACK	2 005/1	
L	7		

TP111010	SIP reference: RFC 3261	[6]		ISUP reference: 127 [1], clause 7.2.3.1.8.9
TSS reference:	SIP-ISUP/Basic call/ Receipt of R	eset circuit m		
	(GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection		3 (,	
criteria:				
ISUP selection				
criteria:				
Test purpose:	GRS for more than one connection	ns received,	a BYE is sent fo	or each connection
	Ensure that the SUT after receiving			
	call association on receipt of a GF		in the confirmed	I state, were the Range and
	Status Parameter value is bigger	than "1":		
	the CLIT shall send a DV	C requests fo	or analy and	aciation
SIP Parameter	the SUT shall send a BY	E requests it	or each can asso	ociation.
values:				
ISUP Parameter				
values:				
Comments:	SIP	s	UT	ISUP
	INVITE(1)	→	→	IAM
		←	←	ACM
		←	←	ANM
	ACK .	→		
		Conve	ersation	
	INVITE(2)	→	→	IAM
	180 Ringing	←	←	ACM
	200 OK INVITE	←	←	ANM
	ACK -	→		
		Conve	ersation	
			←	GRS
	BYE (1)	←	→	GRA
	200 OK BYE	→		
	BYE (2)	←		
	200 OK BYE	→		

TP111011	SIP reference: RFC	3261 [6]		ISUP reference: 027 [1], clause 7.2.3.1.8.9
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection		55 (-	, , , , , , , , , , , , , , , , , , , ,	
criteria:				
ISUP selection criteria:				
Test purpose:	CGB "Hardware failure oriented" for more than one connections received, a BYE is sent for each connection Ensure that the SUT after receiving more than one INVITE sending an IAM message for each			
	call association on receipt of a CGB message in the confirmed state, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" and were the Range and Status Parameter value is bigger than "1": • the SUT shall send a BYE requests for each call association.			
SIP Parameter				
values:				
ISUP Parameter				
values:				
Comments:	SIP		SUT	ISUP
	INVITE(1)	→	→	IAM
	180 Ringing	←	←	ACM
	200 OK INVITE	←	←	ANM
	ACK	→		
		Cor	versation	
	INVITE(2)	→	→	IAM
	180 Ringing	-	←	ACM
	200 OK INVITE	-	←	ANM
	ACK	→		
		Cor	versation	
			←	CGB
	BYE (1)	←	→	CGBA
	200 OK BYE	→		
	BYE (2)	←		
	200 OK BYE	<u>→</u>		

6.2.1.12 Receipt of the Suspend message (SUS) network initiated

Void.

6.2.1.13 Receipt of the Resume message (RES) network initiated

Void.

6.2.2 Interworking from ISUP to SIP

6.2.2.1 Sending of the INVITE message

TP301001	SIP reference: RFC	3261 [6]	ES 283	ISUP reference: 027 [1], clause 7.2.3.2.1.4
TSS reference:	ISUP-SIP /Basic call/Sendin	ISUP-SIP /Basic call/Sending of the INVITE message		
SIP selection				
criteria:				
ISUP selection criteria:				
Test purpose:	IAM contains the complete (Called party numb	er and the send	ing complete indication, sending
root parpood.	of INVITE	sanca party mamo	or and the cond	ing complete indication, conding
	Ensure that the SUT in Idle s			ge containing the complete
	called party number and th	e sending comp	lete indication:	
	0 1 4 10 175			
OID D	Sends the INVITE r	nessage.		
SIP Parameter				
values:				
ISUP Parameter	IAM; Called party number:	with sending com	plete indication	
values:				
Comments:	ISUP/BICC		SUT	SIP
	IAM	→	→	INVITE
	ACM		-	180 Ringing
		Ring	ing tone	
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
	REL	→	→	BYE
	RLC	←	←	200 OK BYE

TP301002	SIP reference: RFC 3261	[6]		ISUP reference:
17301002	SIF reference. RFC 3201	[0]		130F reference. 127 [1], clause 7.2.3.2.1.4
				
TSS reference:	ISUP-SIP /Basic call/Sending of the	ne invite m	essage	
SIP selection				
criteria:				
ISUP selection				
criteria:	1000	<i>c :: :</i> ,	1: 4 4:	
Test purpose:	IAM contains the maximum numb INVITE	er of algits u	sea in the natioi	nai numbering pian, sending of
	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan:			
	sends the INVITE messa	age.		
SIP Parameter				
values:				
ISUP Parameter values:	IAM; Called party number: comp	lete number		
Comments:	ISUP/BICC	S	SUT	SIP
	IAM	→	→	INVITE
	ACM		←	180 Ringing
		Ringi	ng tone	
	ANM	←	+	200 OK INVITE
			→	ACK
		Conve	ersation	
	REL	→	→	BYE
	RLC	(←	200 OK BYE

TP301003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.4
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE m	essage
SIP selection criteria:		
ISUP selection		
criteria:		
Test purpose: SIP Parameter values:	IAM contains a sufficient number of digits to re INVITE Ensure that the SUT in Idle state, on receipt o called party number where the end of addrescalled party number to indicate that a sufficier oute the call to the called party: • sends the INVITE message.	f an IAM message containing the complete as signalling is determined by analysis of the
ISUP Parameter	IAM; Called party number: complete number	
values:		
Comments:	ISUP/BICC S	SUT SIP
	IAM →	→ INVITE
	ACM ←	← 180 Ringing
	Ring	ng tone
	ANM	← 200 OK INVITE
		→ ACK
		ersation
	REL →	→ BYE
	RLC +	← 200 OK BYE

TP301004	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.4	
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message		
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:		of an IAM message with the minimum number of received, by observing the timer Ti/w1 which has	
	expired	cocived, by observing the timer that which has	
	sends the INVITE message.		
SIP Parameter	Ŭ		
values:			
ISUP Parameter			
values:			
Comments:		SUT SIP	
	IAM →		
	T _{I//}	_{/1} expiry	
		→ INVITE	
	ACM ←	← 180 Ringing	
	Ring	ging tone	
	ANM ←	← 200 OK INVITE → ACK	
	Con	versation	
	REL →	→ BYE	
	RLC ←	★ 200 OK BYE	

TP301005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.2	
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE may	essage	
SIP selection		•	
criteria:			
ISUP selection	PICS 1/3		
criteria:			
Test purpose:	IAM received continuity check indicator is set is sent immediately	to "continuity check not required", the INVITE	
	Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter is set to indicate " continuity check not required ":		
	sends a INVITE message.		
SIP Parameter			
values: ISUP Parameter	IAM: Nature of Connection Indicators parameter	tor is not to indicate "continuity shock not	
values:	required"	ter is set to malcate Continuity check not	
Comments:		SUT SIP	
	IAM →	→ INVITE	
	ACM	← 180 Ringing	
	Ringi	ing tone	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Conv	ersation	
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301006	SIP reference: RFC 3261 [6]	ES 283 (ISUP reference: 027 [1], clause 7.2.3.2.1.2
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message		
SIP selection criteria:	NOT PICS 4/11		
ISUP selection criteria:	PICS 4/5 AND PICS 1/3		
Test purpose:	IAM received continuity check indicator is set to "continuity check required on this circuit", the INVITE is sent after COT "successful" is received Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit": • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful".		
values:			
ISUP Parameter	IAM: Nature of Connection Indicators paran	neter which is set t	o "continuity check required
values:	on this circuit" COT: Continuity Indicators parameter "continuity check successful"		
Comments:		•	SIP
Comments:	ISUP/BICC	SUT	SIP
	COT(successful)	→	INVITE
	ACM +	ŕ	180 Ringing
		nging tone	100 Kinging
	ANM ←		200 OK INVITE
		→	ACK
	Co	nversation	
	REL →	→	BYE
	RLC ←	+	200 OK BYE

TP301007	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.2	
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message		
SIP selection	PICS 4/5 AND NOT PICS 4/11		
criteria:			
ISUP selection	PICS 1/3		
criteria:			
Test purpose:	IAM received continuity check indicator is set to "continuity check performed on previous circuit", the INVITE is sent after COT "successful" is received Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check performed on previous circuit": • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful".		
values:			
ISUP Parameter		er which is set to "continuity check performed	
values:	on previous circuit		
Comments:	COT: Continuity Indicators parameter "continuity Indicators parameter "continuity ISUP/BICC	SUT SIP	
Comments.	IAM →	SUI SIF	
	COT(successful)	→ INVITE	
	ACM ←	← 180 Ringing	
		ing tone	
	ANM ←	€ 200 OK INVITE	
		→ ACK	
	Conv	rersation	
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301008	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.2, 7.2.3.2.3
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE n	nessage
SIP selection criteria:	PICS 4/5 AND PICS 4/11	
ISUP selection	PICS 1/3 AND PICS 4/2	
criteria:	1760 1767 1100 472	
Test purpose:	IAM continuity check required received, prec	ondition request in the INVITE
	Ensure that the SUT in Idle state, on receipt of indicator in the Nature of Connection Indicator "continuity check required on this circuit"	•
	SDP offer or answer carrying the co when the Continuity message with t	condition using the SDP offer in the INVITE. The infirmation of a precondition being met is sent the Continuity Indicators parameter set to be received and the requested preconditions are
SIP Parameter	INVITE: Require: precondition	
values:	SDP a=curr:qos local none	
	a=curr:qos remote none	
	a=des:qos mandatory local ser a=des:qos none remote sendre	
	·	
	183: Require: 100rel	
	SDP a=curr:gos local none	
	a=curr:qos remote none a=des:qos mandatory local ser	ndrecy
	a=des:qos mandatory remote s	
	a=conf:qos remote sendrecv	
	UPDATE:	
	SDP a=curr:qos local sendrecv	
	a=curr:qos remote none	
	a=des:qos mandatory local ser a=des:qos mandatory remotes	
	200 OK UPDATE	
	SDP a=curr:qos local sendrecv	
	a=curr:qos remote sendrecv	
	a=des:qos mandatory local ser	
ICUD Devementer	a=des:qos mandatory remote s	
ISUP Parameter values:	on this circuit"	eter which is set to "continuity check required
14.405.	COT: Continuity Indicators parameter "conti	nuity check successful"
Comments:		SUT SIP
	IAM →	→ INVITE
		 183 Session Progress
		→ PRACK
	207	€ 200 OK PRACK
	COT →	→ UPDATE
	ACM ←	← 200 OK UPDATE← 180 Ringing
	, tolvi	→ PRACK
		€ 200 OK PRACK
	Ring	ging tone
	ANM ←	€ 200 OK INVITE
		→ ACK
		versation
	REL →	→ BYE
	RLC +	★ 200 OK BYE

TP301009	SIP ref	erence: RFC 3261 [6]			ISUP reference: 1], clause 7.2.3.2.1.2, 7.2.3.2.3		
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection criteria:	PICS 4/5 AND PICS 4/11						
ISUP selection criteria:	PICS 1/3 AND PICS 4/2						
Test purpose:	IAM continuity INVITE	check performed on a pre	vious c	ircuit received	, precondition request in the		
	indicator in the	e SUT in Idle state, on rece Nature of Connection Indi neck performed on previo	icators	parameter in t	ge where the Continuity Check he IAM is set to indicate		
	SDP when " <i>con</i>	offer or answer carrying the the Continuity message w	e confii vith the	rmation of a pr Continuity Ind	ne SDP offer in the INVITE. The recondition being met is sent icators parameter set to ne requested preconditions are		
SIP Parameter values:	SDP	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv					
	183: Require: 100rel SDP						
	SDP	00 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv					
ISUP Parameter values:	IAM Nature of on previous of	Connection Indicators para	ameter	which is set to	"continuity check performed		
Comments:	ISUP/BICC	y issues parameter of	SU		SIP		
	IAM	→		→	INVITE		
				←	183 Session Progress		
				→	PRACK		
	сот	→		← → ←	200 OK PRACK UPDATE		
	ACM	←		←	200 OK UPDATE 180 Ringing PRACK		
			Б	+	200 OK PRACK		
	ANINA		Ringin	-	200 OK INIVITE		
	ANM	←		←	200 OK INVITE ACK		
		(Conver		,,,,,		
	REL	→		→	BYE		
	RLC	←		←	200 OK BYE		

TP301010	SIP reference: RFC 3261 [6]		ISUP reference: 027 [1], clause 7.2.3.2.1.3			
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE n	nessage				
SIP selection criteria:	PICS 4/20					
ISUP selection criteria:						
Test purpose:	Support of Information Request message (IN Ensure that if no calling party number is rece	ved in the incom				
	sends an INR message to request the calling party number and not sends the INVITE request until receiving an INF message with calling party number. If no calling party number is received in the INF message, O-MGCF may reject or continue the call based on local configuration.					
SIP Parameter						
values:						
ISUP Parameter						
values:						
Comments:		SUT	SIP			
	IAM →					
	INR +					
	INF →	_				
		→	INVITE			
	ACM ←	+	180 Ringing			
	Ringing tone	_	000 OK NIVITE			
	ANM ←	← →	200 OK INVITE ACK			
	Conversation					
	REL →	→	BYE			
	RLC ←		200 OK BYE			

TP301011	SIP reference: RFC 3261 [6]		ISUP reference: 3 027 [1], clause 7.2.3.2.1a					
TSS	ISUP-SIP/Basic call/Sending of the INVITE message							
reference:								
SIP selection								
criteria:								
selection								
criteria:								
Test	Sending of INVITE without determining to	he end of address s	signalling					
purpose:	Ensure that if the O-MGCF sends an INVITE request before the end of address signalling is determined, the O-MGCF shall: - start timer Ti/w2; and - be prepared to process SAM - be prepared to handle incoming SIP 404 or 484 error. On receipt of a SAM from the ISUP side, the O-MGCF shall: stop timer Ti/w3 (if it is running); send an INVITE request complying to the following:							
	restart Ti/w2.	- The INVITE request shall include all digits received so far for this call in the Request-URI.						
SIP	restait 1/wz.							
Parameter								
values:								
ISUP								
Parameter								
values:	IOUR/DIGG	OUT	OID					
Comments:	ISUP/BICC IAM →	SUT	SIP					
	SAM ->	÷	INVITE 404/484 ACK INVITE 404/484 ACK					
	SAM ACM ← ANM	Ringing tone	INVITE 180 Ringing 200 OK INVITE					
	REL → ←	Conversation	▶ BYE					

2.3.2.2.2
ansmission
with the "a="

TP301013	SIP reference: RFC 3261 [6]		ISUP reference:					
TSS reference:	ES 283 027 [1], clause 7.2.3.2.2.2							
SIP selection	ISUP-SIP/Basic call/ Sending of the INVITE message							
criteria:	Based on table 9							
ISUP selection								
criteria:								
	Manning of LICI into the CDD in the cont	N // // // // // // // // // // // // //						
Test purpose:	Mapping of USI into the SDP in the sent	INVITE						
	Ensure that the SUT in the Idle state on rinformation parameter set to USI_VALU sends an INVITE message, with and "m=" lines set to a_b_m_LII	the media descript	-					
SIP Parameter	INVITE: a_b_m_LINE_VALUE	IL_V/\LOL.						
values:	INVITE. a_b_m_enve_vacoe							
ISUP Parameter values:	IAM: USI: ISUP_USI							
Comments:	ISUP/BICC	SUT	SIP					
	IAM →	→	INVITE					
	ACM ←	←	180 Ringing					
	R	nging tone						
	ANM ←	+	200 OK INVITE					
		→	ACK					
	Co	nversation						
	REL →	→	BYE					
	RLC ←	←	200 OK BYE					

P301014	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.2							
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message								
SIP selection criteria:	PICS 2/8								
ISUP selection criteria:									
Test purpose:	SDP offer using the AMR codec								
	Ensure that the SUT in the Idle state on rece call, with the user information parameter see sends an INVITE message, with the	et to USI_VALUE:							
SIP Parameter	Offer: m=audio RTP/AVP dynimac PT								
values:	a = rtpmap dynimac PT AMR Answer: m=audio RTP/AVP dynimac PT a = rtpmap dynimac PT AMR								
ISUP Parameter values:	IAM: USI= USI_VALUE (PIXIT)								
Comments:	ISUP/BICC S	UT SIP							
	IAM →	→ INVITE							
	ACM ←	← 180 Ringing							
	Ringir	ng tone							
	ANM ←	← 200 OK INVITE							
		→ ACK							
	Conve	ersation							
	REL →	→ BYE							
	RLC +	← 200 OK BYE							

TP301015	SIP reference: I	RFC 3261 [6]		ISUP reference:			
			ES 283 ()27 [1], clause 7.2.3.2.2.2			
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message						
SIP selection	NOT PICS 2/8						
criteria:							
ISUP selection							
criteria:							
Test purpose:	No AMR codec in the S.	DP, when no equipme	nt implements	the AMR codec			
	Ensure that the SUT in t	the Idle state on receip	ot of an IAM m	essage indicating a speech			
	call, with the user information parameter set to USI_VALUE and the IMS network serves						
	that no user equipment		codec, then the	e AMR codec shall be			
	excluded from the SDP	offer.					
SIP Parameter	INVITE: SDP no AMR c	odec					
values:							
ISUP Parameter	IAM: USI= USI_VALUE	(PIXIT)					
values:							
Comments:	ISUP/BICC	SU	T	SIP			
	IAM	→	→	INVITE			
	ACM	←	←	180 Ringing			
		Ringing	tone				
	ANM	←	←	200 OK INVITE			
			→	ACK			
		Conver	sation	-			
	REL	→	→	BYE			
	RLC	,	ŕ	200 OK BYE			
	INLO			200 ON DIL			

Table 8

	Values for test purposes TP301012									
	ISUP			SDP - a_b_n	_LINE_VALU	Ε				
	TMR parameter		m= l	ine	b= line	a= line				
	TMR codes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>: <bandwidt h-value></bandwidt </modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt>				
VA_01	"speech"	Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)				
VA_02	"speech"	Audio	RTP/AVP	Dynamic PT (and possibly a second Dynamic PT)	AS:64	rtpmap: <dynamic-pt> PCMU/8000 (and possibly rtpmap:<dynamic-pt> PCMA/8000)</dynamic-pt></dynamic-pt>				
VA_03	"speech"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000				
VA_04	"speech"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>				
VA_05	"3,1 KHz audio"	Audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000				
VA_06	"3,1 KHz audio"	Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)				
VA_07	"3,1 KHz audio"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000				
VA_08	"64 kbit/s unrestricted"	Audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000				
VA_9	"64 kbit/s unrestricted"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>				

Table 9

	1			ues for test pur	poses 1P3	01012, 1P30					
VA		ISU		1	SDP - a_b_m_LINE_VA						
TMR		USI para	meter	HLC IE in ATP	m= line			m= line b= line		b= line	a= line
	TMR	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristi cs Identification	<media></media>	<transport< th=""><th><fmt-list></fmt-list></th><th><modifier>: <bandwidth -value></bandwidth </modifier></th><th>rtpmap:<dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt></th></transport<>	<fmt-list></fmt-list>	<modifier>: <bandwidth -value></bandwidth </modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt>		
VA_01	"speech"	"Speech"	"G.711 μ-law"	Ignore	audio	RTP/AVP	0 (and possibly 8) (NOTE 1)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) (see note 1)		
VA_02	"speech"	"Speech"	"G.711 μ-law"	Ignore	audio	RTP/AVP	Dynamic PT (and possibly a second Dynamic PT) (NOTE 1)	AS:64	rtpmap: <dynamic-pt> PCMU/8000 (and possibly rtpmap:<dynamic- PT> PCMA/8000) (see note 1)</dynamic- </dynamic-pt>		
VA_03	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000		
VA_04	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>		
VA_05	"3,1 kHz audio"	USI Absent		Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000		
VA_06	"3,1 kHz audio"	"3,1 kHz audio"	"G.711 μ-law"	(NOTE 3)	audio	RTP/AVP	0 (and possibly 8) (NOTE 1)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) (see note 1)		
VA_07	"3,1 kHz audio"	"3,1 kHz audio"	"G.711 A-law"	(NOTE 3)	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000		
VA_08	"3,1 kHz audio"	"3,1 kHz audio"		"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on ITU-T Recommendation T.38 [31].		
VA_09	"3,1 kHz audio"	"3,1 kHz audio"		"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on ITU-T Recommendation T.38 [31]		
VA_10	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000		
VA_11	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000 (see note 2)</dynamic-pt>		

NOTE 1 Both PCMA and PCMU could be required.

NOTE 2 CLEARMODE is specified in RFC 4040 [19].

NOTE 3 HLC is normally absent in this case. It is possible for HLC to be present with the value "Telephony", although 6.3.1/Q.939 indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

TP301016	SIP reference	RFC 3261 [6]		2 202	ISUP reference:				
T00 (ES 283 027 [1], clause 7.2.3.2.2.2								
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message								
SIP selection	PICS 2/8								
criteria:									
ISUP selection									
criteria:									
Test purpose:	The SUT defines the F	RS and RR ban	dwidth modifiers	whe	en AMR codec is used				
	Ensure that the if the 0	D-MGCF detern	nines that a spec	ech c	all is incoming, the O-MGCF				
					S bandwidth modifiers specified				
	in RFC 3556 [17] to disable RTCP.								
SIP Parameter	INVITE: SDP b=RS: <i< th=""><th>bandwidth-valu</th><th>e></th><th></th><th></th></i<>	bandwidth-valu	e>						
values:	b=RR:<	bandwidth-valu	e>						
ISUP Parameter									
values:									
Comments:	ISUP/BICC		SUT		SIP				
	IAM	→		→	INVITE				
	ACM	←		←	180 Ringing				
			Ringing tone		3 3				
	ANM	←	runging tono	←	200 OK INVITE				
	7 (1 414)	`		→	ACK				
			Camuramantian	7	AUN				
	551	_	Conversation		5)/5				
	REL	→		→	BYE				
	RLC	<u> </u>		←	200 OK BYE				

TP301017	SIP reference: RFC 3261 [6]			ISUP reference:					
			ES 283 ()27 [1], clause 7.2.3.2.2.1					
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message								
SIP selection									
criteria:									
ISUP selection									
criteria:									
Test purpose:	Mapping of Called party number into to	he To heade	er user=ph	one is included					
	Ensure that the SUT is mapping the C		address in	formation contained in the					
	Called Party Number parameter of the	IAM:							
	to the addr-spec component of the state								
	"user=phone" URI parameter	if the To he	ader field	contains a sip: URI.					
SIP Parameter	INVITE: To: sip:; user=phone								
values:									
ISUP Parameter									
values:									
Comments:	ISUP/BICC	SUT		SIP					
	IAM →		→	INVITE					
	ACM ←		←	180 Ringing					
		Ringing tor	ne						
	ANM ←		←	200 OK INVITE					
			→	ACK					
		Conversati	on						
	REL →		→	BYE					
	RLC ←		←	200 OK BYE					

TP301018	SIP reference: RFC 3261 [6]			ISUP reference: 27 [1], clause 7.2.3.2.2.1				
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message							
SIP selection								
criteria:								
ISUP selection								
criteria:								
Test purpose:	Mapping of Called party number into	the To hea	der received	digits in the addr-spc				
	component							
	Engure that the SLIT is mapping the C	Collad Darts	, addraga inf	formation contained in the				
	Ensure that the SUT is mapping the C							
	Called Party Number parameter of the IAM and the and the followed SAM:							
	to the addr-spec component	of the To I	neader field					
SIP Parameter	INVITE: To:							
values:								
ISUP Parameter								
values:								
Comments:	ISUP/BICC	SUT		SIP				
	IAM →		→	INVITE				
	ACM ←		←	180 Ringing				
		Ringing t	one					
	ANM		←	200 OK INVITE				
			→	ACK				
		Conversa						
	REL →		→	BYE				
	RLC +		-	200 OK BYE				

SIP reference: RFC	3261 [6]			ISUP reference: 027 [1], clause 7.2.3.2.2.1	
ISUP-SIP/Basic call/ Sending of the INVITE message					
Mapping of Called party number into the To header as a SIP URI					
Ensure that the SUT is mapping in the Called Party Number parameter contained in the Called Party address information of the IAM and followed SAM: • to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.					
INVITE: To: sip:; user=p	ohone				
ISUP/BICC		SUT		SIP	
IAM	→		→	INVITE	
			←	404/484	
			→	ACK	
SAM	→		→	INVITE	
ACM	←	Ringing tone	←	180 Ringing	
ANM	←	. anging tone	←	200 OK INVITE ACK	
		Conversation	-		
REL	→	230.00.1011	→	BYE	
	-		←	200 OK BYE	
	ISUP-SIP/Basic call/ Sending Mapping of Called party num Ensure that the SUT is mapp Called Party address informa to the addr-spec cor "user=phone" URI p INVITE: To: sip:; user=p ISUP/BICC IAM SAM ACM	ISUP-SIP/Basic call/ Sending of the Mapping of Called party number into Ensure that the SUT is mapping in the Called Party address information of t • to the addr-spec componen "user=phone" URI paramete INVITE: To: sip:; user=phone ISUP/BICC IAM → SAM → ACM ← ANM ← REL	ISUP-SIP/Basic call/ Sending of the INVITE message Mapping of Called party number into the To header at Ensure that the SUT is mapping in the Called Party N Called Party address information of the IAM and follow • to the addr-spec component of the To header "user=phone" URI parameter if the To header "user=phone" URI parameter if the To header "user=phone" ISUP/BICC SUT IAM → SAM → ACM ← Ringing tone ← Conversation REL	ISUP-SIP/Basic call/ Sending of the INVITE message	

TP301020	SIP reference: RFC 3261	[6]		ISUP reference: 027 [1], clause 7.2.3.2.2.4		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Initial Address message (IAM)/					
SIP selection criteria:						
ISUP selection criteria:	PICS 4/5					
Test purpose:	Ensure that if the Hop Counter procedure is supported in the CS network, the O-MGCF shall use the Hop Counter parameter to derive the Max-Forwards SIP header. Due to the different default values (that are based on network demands/provisions) of the SIP Max-Forwards header and the Hop Counter, an adaptation mechanism shall be used to adopt the Hop Counter to the Max Forwards at the O-MGCF. Max-Forwards for a given message should be monotone decreasing with each successive visit to a SIP entity, regardless of intervening interworking, and similarly for Hop Counter.					
SIP Parameter values:						
ISUP Parameter values:						
Comments:	ISUP/BICC	SL	IT	SIP		
	IAM →		→	INVITE		
	ACM ←		←	180 Ringing		
		Ringing	g tone			
	ANM ←		← →	200 OK INVITE ACK		
		Conver	sation			
	REL →		→	BYE		
	RLC ←		<u></u>	200 OK BYE		

Table 10: Hop counter-Max forwards

Н	op Counter	= X	Max-Forwards	= Y = Integer part of (X * Factor)
NOTE:	The Mapping of va	alue X to Y should be done w	ith the used (implen	nented) adaptation mechanism.

TP301021	SIP reference: RFC 3261 [6]	ES 283	ISUP reference: 027 [1], clause 7.2.3.2.2.1
TSS reference:	ISUP-SIP/Basic call/ Sending of the INV		<u> </u>
SIP selection criteria:	•		
ISUP selection criteria:	PICS 1/6		
Test purpose:	Mapping of a "international number" into	the To header and	Request URI
SIP Parameter	Ensure that the called party number par Request URI and To header of the INVI "user=phone" it shall contain an Internata "+" sign (e.g. tel:+4911231234567). If the Request URI is a sip URI with "use telecommunication number prefixed by sip:+4911231234567@host). Ensure that the SUT is mapping the Cal Called Party Number parameter, Nature IAM: • to the format of the To header Addess signal. INVITE: To:, Request URI	TE Request. If the Fional public telecomer=phone" it shall coa" +" sign and a hos led Party address in for address = "Interest and a dot a ddress = "Interest and a ddress = "Interest and a ddress = "Interest and a ddress a ddress and a ddress a ddress and a ddress a ddress and	Request URI is a tel URI with imunication number prefixed by intain an International public t portion (e.g. formation contained in the ernational number" of the
values:			
ISUP Parameter			
values: Comments:	ISUP/BICC	SUT	SIP
Comments.	IAM →	→	INVITE
	ACM ←	<u>+</u>	180 Ringing
	-	Ringing tone	
	ANM ←	₩gg toe	200 OK INVITE
		→	ACK
		Conversation	
	REL →	→	BYE
	RLC ←	+	200 OK BYE

TP301022	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.1	
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVIT		
SIP selection		· ·	
criteria:			
ISUP selection	NOT PICS 1/6		
criteria:			
Test purpose:	Mapping of Called party number "national (significant) number" into the To header and Request URI user=phone is included		
	Request URI of the INVITE Request. The shall contain an International public telecotel:+4911231234567). If the Request URI is a sip URI with "usertelecommunication number prefixed by a "sip:+4911231234567@host). Ensure that the SUT is mapping the Called Party Number parameter, Nature o of the IAM:	d Party address information contained in the f address = "National (significant) number" Id and Request URI inserting "+" CC before the	
SIP Parameter	INVITE: To:		
values:			
ISUP Parameter values:			
Comments:	ISUP/BICC	SUT SIP	
Johnnents.	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
		iging tone	
	ANM ←	€ 200 OK INVITE	
		→ ACK	
	Cor	nversation	
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301023	SIP reference: RFC 326	1 [6]		ISUP reference:
)27 [1], clause 7.2.3.2.2.7
TSS reference:	ISUP-SIP/Basic call/ Sending of	the INVITE m	essage	
SIP selection	PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Mapping of USI parameter into F	PSTN XML Be	earerCapability	•
	OUT:			
	Ensure that the SUT in Idle state		r an IAM mess	age User Service
	Information (USI) set to USI_VA	ALUE:		
	1 d 1819/175	201 - 01	DOTAL VIAL B	0 (00)
		age with the	PSIN XML BE	earer Capability (BC) set to
010.0	USI_VALUE.	(5.0)		NAT)
SIP Parameter	INVITE; PSTN XML BearerCapa	ability (BC): (JSI_VALUE (F	IXII)
values:				
ISUP Parameter	IAM; USI : USI_VALUE (PIXIT)			
values:				
Comments:	ISUP/BICC	SU	T	SIP
		→	→	INVITE
	ACM	-	←	180 Ringing
		Ringing	g tone	
	ANM	+	←	200 OK INVITE
			→	ACK
		Conver	sation	
	REL =	→	→	BYE
	RLC	-	+	200 OK BYE

TP301024	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.7	
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE	= =:	
SIP selection	PICS 4/18	3	
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of USI and USI prime parameter into PSTN XML BearerCapability Ensure that the SUT in Idle state, on receipt of an IAM message USI set to "speech" and		
	sends the INVITE message with the "speech" (the USI value) and the se "unrestricted digital information with		
SIP Parameter	value).	anacah	
values:	INVITE; first PSTN XML Bearer Capability: speech second PSTN XML Bearer Capability: unrestricted digital information with		
values.	tones and announcements	y. unrestricted digital information with	
ISUP Parameter	IAM; USI: speech		
values:	USI Prime: unrestricted digital information w	th tones and announcement	
Comments:		UT SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringii	ng tone	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Conve	ersation	
	REL →	→ BYE	
	RLC ←	★ 200 OK BYE	

P301025	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8	
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE r	nessage	
SIP selection criteria:	PICS 4/18	•	
ISUP selection criteria:			
Test purpose:	Mapping of FCI "ISUP not used all the way" is	nto PSTN XML ProgressIndicator #1	
		of an IAM message containing the ISUP TN XML ProgressIndicator set to "call is not so information is available in-band (#1)".	
SIP Parameter	INVITE;		
values:	PSTN XML ProgressIndicator: call is not end-to-end ISDN: further call progress		
	information is available in-band (#1)		
ISUP Parameter	IAM; ISUP indicator: ISUP not used all the way		
values:			
Comments:	ISUP/BICC S	UT SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringir	ng tone	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Conve	rsation	
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301026	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.2.2.8	
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE m	essage	
SIP selection	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of FCI "Originating access non ISDN	" into PSTN XML ProgressIndicator #3	
	English that the CLIT is talle atotal as a receipt of	f on IAM managers containing the ICUD	
	Ensure that the SUT in Idle state, on receipt o indicator set to "ISUP used all the way" and t		
	access non-ISDN":	The ISDN access indicator set to originating	
	access non-iodiv .		
	• sands the INVITE massage with the	PSTN XML ProgressIndicator se tot	
	"Originating access is non ISDN (#3)		
SIP Parameter	INVITE:		
values:	PSTN XML ProgressIndicator: Originating access is non ISDN (#3)		
ISUP Parameter	IAM:		
values:	ISUP indicator: ISUP used all the way		
values.	ISDN access indicator: originating access non-ISDN		
Comments:	ISUP/BICC SL		
O O I I I I I I I I I I I I I I I I I I	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	1		
	Ringing	€ 200 OK INVITE	
	ANM ←		
		→ ACK	
	Convei		
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301027	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8	
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE		
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:			
Test purpose:	Mapping of FCI "ISUP used all the way" and "Originating access non ISDN" into PSTN XML ProgressIndicator #3		
	Ensure that the SUT in Idle state, on receipt	of an IAM message containing the ISUP I the ISDN access indicator set to "originating"	
	access non-ISDN",	Title 13DN access indicator set to originating	
	,	e PSTN XML ProgressIndicatorPSTN XML	
	ProgressIndicator set to "Originati		
SIP Parameter	INVITE; PSTN XML ProgressIndicatorPSTN XML ProgressIndicator: Originating		
values:	access id non ISDN (#3)	•	
ISUP Parameter	IAM; ISUP indicator: ISUP used all the way		
values:	ISDN access indicator: originating access r	non-ISDN	
Comments:	ISUP/BICC S	SUT SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringi	ng tone	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Conve	ersation	
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301028	SIP reference: RFC 3261 [6]		ISUP reference:
			027 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVI	TE message	
SIP selection	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of FCI "ISUP used all the way" a ProgressIndicator #6 Ensure that the SUT in Idle state, on receindicator set to "ISUP used all the way" access ISDN" • sends the INVITE message with "originating access ISDN" (#6). INVITE; PSTN XML ProgressIndicator:	eipt of an IAM mess and the ISDN acce the PSTN XML P i	sage containing the ISUP ss indicator set to "originating rogressIndicator set to
values:			
ISUP Parameter	IAM; ISUP indicator: ISUP used all the way		
values:	ISDN access indicator: originating acce		
Comments:	ISUP/BICC	SUT	SIP
	IAM →	→	INVITE
	ACM ←	←	180 Ringing
	Ri	inging tone	
	ANM ←	←	200 OK INVITE
		→	ACK
	Co	onversation	
	REL →	→	BYE
	RLC ←	+	200 OK BYE

TP301029	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8	
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE n	nessage	
SIP selection	PICS 4/18	•	
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of FCI "ISUP used all the way" and		
	a Progress Indicator into PSTN XML Progress	sIndicator #6	
	Figure that the CLIT is talled at the congression to	of an IAM manager containing the ICUD	
	Ensure that the SUT in Idle state, on receipt of	the ISDN access indicator set to "originating	
		meter (ATP) containing progress indicator set	
	to PI VALUE,	meter (ATT) containing progress indicator set	
	 sends the INVITE message with the 	PSTN XML ProgressIndicator set to	
	"originating access ISDN" (#6) and F		
SIP Parameter	INVITE; PSTN XML ProgressIndicator: "originating access ISDN" (#6) and PSTN XML		
values:	ProgressIndicator: PI_VALUE (PIXIT)		
ISUP Parameter	IAM; ISUP indicator: ISUP used all the way		
values:	ISDN access indicator: originating access ISDN		
	ATP progress indicator: PI_VALUE (PIXIT)		
Comments:		UT SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	_	ig tone	
	ANM ←	← 200 OK INVITE	
	_	→ ACK	
		rsation	
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301030	SIP reference: RFC 3261 [6]	50.0	ISUP reference:
			33 027 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Sending of the IN\	/ITE message	
SIP selection	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of ATP contains a LLC into a F	PSTN XML LLC ii	n the sent INVITE
	Ensure that the SUT in Idle state, on rec Transport Parameter (ATP) containing t LLC_VALUE, • sends the INVITE message windle LLC VALUE.	he Low Layer C	
SIP Parameter values:	the PSTN XML LowLayerCompatibility: LLC_VALUE (PIXIT)		
ISUP Parameter	IAM; ATP LLC: LLC_VALUE (PIXIT)		
values:			
Comments:	ISUP/BICC	SUT	SIP
	IAM →		→ INVITE
	ACM ←		← 180 Ringing
	I	Ringing tone	
	ANM		← 200 OK INVITE
			→ ACK
		Conversation	
	REL →		→ BYE
	RLC ←		€ 200 OK BYE

TP301031	SIP reference: RFC 3261 [6]		ISUP reference:
11 301031	on reference. Ki o 3201 [0]	FS 283	027 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Sending of the IN		027 [1], 010000 71210121210
SIP selection	PICS 4/18	711L mossage	
criteria:	1 100 4/10		
ISUP selection			
criteria:			
Test purpose:	Mapping of ATP contains a HLC into a	PSTN XML HLC in	the sent INVITE
	HLC_VALUE.	the High Layer Conth	mpatibility (HLC) set to
SIP Parameter values:	INVITE: PSTN XML HighLayerCompa	tibility: HLC_VALU	IE (PIXIT)
ISUP Parameter values:	IAM; ATP HLC: HLC_VALUE (PIXIT)		
Comments:	ISUP/BICC	SUT	SIP
	IAM →	-)	INVITE
	ACM ←	+	180 Ringing
		Ringing tone	
	ANM ←	+	200 OK INVITE
)	ACK
		Conversation	
	REL →)	BYE
	RLC ←	+	200 OK BYE

TP301032	SIP reference: RFC 3261 [6]	ISUP reference:
17301032	Sir felerence. Krc 3201 [0]	
T00 (IOUE OID : II/O I: (II INIVITE	ES 283 027 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE m	nessage
SIP selection	PICS 4/18	
criteria:		
ISUP selection		
criteria:		
Test purpose:	Mapping of User Teleservice Information para INVITE	ameter into a PSTN XML HLC in the sent
	Ensure that the SUT in Idle state, on receipt of Teleservice containing the High Layer Comp	atibility (HLC) set to HLC_VALUE,
	Sends the INVITE message with the HLC_VALUE.	PSTN XML HighLayerCompatibility set to
SIP Parameter	INVITE: PSTN XML HighLayerCompatibility	<i>y</i> : HLC_VALUE (PIXIT)
values:		
ISUP Parameter	IAM; User Teleservice Information: HLC_VALUE (PIXIT)	
values:		
Comments:	ISUP/BICC SU	JT SIP
	IAM →	→ INVITE
	ACM ←	← 180 Ringing
	Ringin	a tone
	ANM ←	€ 200 OK INVITE
		→ ACK
	Conve	rsation
	REL →	→ BYF
		2 2.2
	RLC C	€ 200 OK BYE

TP301033	SIP reference: RFC 3261 [6]		ISUP reference: 27 [1], clause 7.2.3.2.2.8		
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVIT	message			
SIP selection	PICS 4/18				
criteria:					
ISUP selection					
criteria:					
Test purpose:	Mapping of two HLC parameter contained in an ATP parameter into two PSTN XML HighLayerCompatibility elements Ensure that the SUT in Idle state, on receipt of an IAM message containing the Access				
	Transport Parameter (ATP) containing two High Layer Compatibility (HLC) set to respectively HLC_VALUE1 and HLC_VALUE2, • sends the INVITE message with two PSTN XML HighLayerCompatibility in the same order HLC_VALUE1 and HLC_VALUE2.				
SIP Parameter values:	INVITE; first PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT) second PSTN XML HighLayerCompatibility: HLC_VALUE2 (PIXIT)				
ISUP Parameter	IAM;				
values:	ATP first HLC: HLC_VALUE1 (PIXIT) ATP second HLC: HLC_VALUE2 (PIXIT)				
Comments:	ISUP/BICC	SUT	SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	Rin	ging tone			
	ANM ←	←	200 OK INVITE		
		→	ACK		
		versation			
	REL →	→	BYE		
	RLC ←		200 OK BYE		

TP301034	SIP reference: RF	C 3261 [6]		ISUP reference: 027 [1], clause 7.2.3.2.2.8	
TSS reference:	ISUP-SIP/Basic call/ Send	ing of the INVITE n		. [.],	
SIP selection criteria:		3			
ISUP selection criteria:					
Test purpose:	Mapping of calling party ca	ategory into cpc pai	rameter in the I	P-Aserted-Identity	
	Ensure that the SUT map the calling party category ISUP_CPC into the cpc SIP_CPC parameter in the P-Asserted-Identity and Accept-Contact header parameter "language" SIP_LANG.				
SIP Parameter	INVITE; P-Asserted-Identit	ty, Accept-Contact			
values:					
ISUP Parameter	IAM; Calling party category	У			
values:					
Comments:	ISUP/BICC	SI	JT	SIP	
	IAM	→	→	INVITE	
	ACM	←	←	180 Ringing	
		Ringin	g tone	5 5	
	ANM	←	ັ ←	200 OK INVITE	
			→	ACK	
		Conve	rsation		
	REL	→	→	BYE	
	RLC	+	<u></u>	200 OK BYE	

Values for test purposes TP301034					
ISUP_CPC	SIP Parameters				
received calling party's category	SIP_CPC in P-Asserted-Identity	SIP_LANG Accept- Contact 'language'			
operator, language French	operator	French			
operator, language English	operator	English			
operator, language German	operator	German			
operator, language Russian	operator	Russian			
operator, language Spanish	operator	Spanish			
ordinary calling subscriber	ordinary				
test call	test				
payphone	payphone				
mobile terminal located in the home PLMN	cellular				
mobile terminal located in a visited PLMN	cellular roaming				
IEPS call marking for preferential call set up	ieps				

6.2.2.2 Receipt of the SAM message after INVITE has been send

TP302001	SIP reference	ce: RFC 3261 [6]		ISUP reference:
				ES 283 027 [1], clause 7.2.3.2.1.4
TSS reference:	ISUP-SIP/Basic call/F	Receipt of SAM after INVIT	E ha	as been sent
SIP selection	PICS 3/1			
criteria:				
ISUP selection	PICS 3/5 AND NOT F	PICS 3/8		
criteria:				
Test purpose:	Overlap procedure no	ot supported, SAM is ingno	ored	
	Ensure if the SUT is supporting en bloc addressing towards the SIP network, subsequent			
	SAMs received after	the SUT has sent the INVI	TE a	re ignored.
SIP Parameter				
values:				
ISUP Parameter	SAM; subsequent n	umber (PIXIT)		
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	SAM	→		
	ACM	←	←	180 Ringing
	Ring	ging tone		
	ANM	←	←	200 OK INVITE
			→	ACK
	Conversation			1
		→	→	BYE
	RLC	(←	200 OK BYE

TP302002	SIP re	ference: RFC 3261 [6]		ISUP reference:		
11 002002		.o.ooo.	ES	283 027 [1], clauses 7.2.3.2.1a and 7.2.3.2.1.4		
TSS reference:	ISUP-SIP/Bas	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent				
SIP selection criteria:	PICS 3/2	·				
ISUP selection criteria:	PICS 3/8					
Test purpose:	Overlap procedure supported by determining the end of address signalling. sending complete indication received Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "continuity check not required" on receipt of a SAM containing the complete called party number and the sending complete indication and the SUT On receipt of a SAM from the ISUP the SUT shall: Stop timer TOIW3 (if it is running). TOIW2 shall be restarted and the SUT shall invoke the following procedures:					
SIP Parameter	 a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-MGCF may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 					
values:						
ISUP Parameter						
values:						
Comments:	ISUP/BICC	SU	Т	SIP		
	IAM	→	→	INVITE		
			←	404/484		
			→	ACK		
	SAM	→	→	INVITE		
			←	404/484		
			→	ACK		
	SAM	→	→ ←	INVITE 404/484		
			→	ACK		
	CAN(E)	_\$	•	IND/ITE		
	SAM(F)	7	→	INVITE		
	ACM	←	←	180 Ringing		
		Ringing tone				
	ANM	←	←	200 OK INVITE		
			→	ACK		
			Conversation			
		→	→	BYE		
	RLC	,	÷	200 OK BYE		
<u> </u>	IVEO			200 OR DIL		

TP302003	SIP ref	erence: RFC 3261	[6]		ISUP reference:	
				ES	283 027 [1], clauses 7.2.3.2.1a and 7.2.3.2.1.4	
TSS reference:	ISUP-SIP/Basi	ic call/Receipt of SA	M after invit	e has be		
SIP selection criteria:	PICS 3/2					
ISUP selection criteria:	PICS 3/8					
criteria: Test purpose:	ensure that the indicator in the check not requational number sends an INVI Stop timer TOI TOIW2 shall be a) The Rereceived so b) A new I INVITE is second to the resource reserve	in the national number of the national number of SUT in Idle state, Nature of Connection of the state of the	on receipt of on Indicator a SAM and ched, the SI ning all digits SUT shall in To header fine Call-ID ar ain a new Sybeen rese	f an IAM s param the max UT s receive voke the eld of the DP offer rved for	of address signalling. Maximum number need message containing Continuity Check eter which is set to indicate "continuity kimum number of digits used in the ed in the IAM and the SAM(s). e following procedures: e new INVITE shall contain all digits header (including tag) as the previous The O-MGCF may re-use any this call. This re-use of existing e precondition attributes for the SDP	
CID Deventor	d) All othe	parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM.				
SIP Parameter values:						
ISUP Parameter						
values:	ISUP/BICC		SUT		SIP	
Comments:	IAM	→	301	→	INVITE	
	IAW	7		-	404/484	
				→	ACK	
	SAM	→		→	INVITE	
	OAW	-		÷	404/484	
				→	ACK	
	SAM	→		→	INVITE	
	OAW	-		ŕ	404/484	
				→	ACK	
	SAM	→		→	INVITE	
	ACM	← Ringing tone		←	180 Ringing	
	ANM	Kinging tone ←		← →	200 OK INVITE ACK	
			Conv	ersation		
		→		→	BYE	
	RLC	+		←	200 OK BYE	

TP302004	SIP refe	erence: RFC 3261	[6]		ISUP reference:
11 302004	Oil Tele	sterice. IXI O 3201	[0]	ES	283 027 [1], clauses 7.2.3.2.1a and
					7.2.3.2.1.4
TSS reference:	ISUP-SIP/Basic	call/Receipt of SA	M after invit	e has be	een sent
SIP selection	PICS 3/2				
criteria:					
ISUP selection criteria:	PICS 3/8				
Test purpose:					of address signalling. Sufficient number
	of digits has be	en received to rout	e the call re	ceived	
	indicator in the check not requ	Nature of Connecti	ion Indicator a SAM and	s param the suff	message containing Continuity Check eter which is set to indicate "continuity icient number of digits has been
	sends an INVIT	E message contair	ning all digits	s receive	ed in the IAM and the SAM(s).
	Stop timer TOIV	V3 (if it is running).			
				voke the	e following procedures:
			To header fie	eld of the	e new INVITE shall contain all digits
		far for this call.	0 11 10	. –	
	b) A new INVITE is se		ne Call-ID ar	nd From	header (including tag) as the previous
			ain a new Sl	DP offer	. The O-MGCF may re-use any
					this call. This re-use of existing
					precondition attributes for the SDP
	parameters	in question.			
			w INVITE ar	e interw	orked from the parameters of the
	original IAM	•			
SIP Parameter					
values:					
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
Comments.	IAM	→	551	→	INVITE
	IAW	•		É	404/484
				÷	ACK
				-	non
	SAM	→		→	INVITE
				←	404/484
				→	ACK
		_		_	
	SAM	→		→	INVITE
				(404/484
				→	ACK
	SAM	→		→	INVITE
	ACM	←		←	180 Ringing
		Ringing tone			
	ANM	←		←	200 OK INVITE
				→	ACK
			Conv	ersation	
		→		→	BYE
	RLC	+		+	200 OK BYE

TP302005	SIP reference: RF	C 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.2.4
TSS reference:	ISUP-SIP/Basic call/Recei	ipt of SAM after inv	ite has be	een sent
SIP selection criteria:	NOT PICS 3/2			
ISUP selection criteria:	PICS 3/8			
Test purpose:		,		of address signalling. Ti/w1 is expired message containing Continuity Check
	Ensure that the SUT in Idle state, on receipt of an IAM message containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "continuity check not required" on receipt of a SAM start Ti/w1. After Ti/w1 is expired, the SUT: • sends an INVITE message containing all digits received in the IAM and the SAM.			
SIP Parameter	- condo an nvine n	loodago cornaming	an aigito	received in the 17 twi drid the 67 twi.
values:				
ISUP Parameter				
values:				
Comments:	ISUP/BICC	SUT		SIP
	.,	→ Start T _{i/w1}		
	SAM	→ Start T _{i/w1}		
	SAM	→ Start T _{i/w1}		
	SAM	→ Start T _{i/w1}		
		T _{i/w1} expired	→	INVITE
	ACM(no indication)	←		
		-	←	180 Ringing
	Ringing	tone		3 3
		←	←	200 OK INVITE
			→	ACK
		Con	versation	
		→	→	BYE
	RLC	-	-	200 OK BYE

TP302006	SIP re	ference: RFC 3261 [6]	E	ISUP reference: ES 283 027 [1], clause 7.2.3.2.1a
TSS reference:		sic call/Receipt of SAM after invi	te has be	een sent
SIP selection criteria:	PICS 3/2			
ISUP selection criteria:	PICS 3/9			
Test purpose:	Overlap proce	edure supported without determi	ning the	end of address signalling
	Check indicate			message containing the Continuity parameter which is set to indicate
	• Send	ls an INVITE message start Ti/w	/1 and Ti	i/w2.
	• On re	eceipt of a 404/484 the SUT sha	ll send a	ACK, stop Ti/w2 and start Ti/w3:
	 On receipt of a SAM from the ISUP the SUT shall send an INVITE request: a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) All other contents of the new INVITE are interworked from the parameters of the original IAM. c) Start Ti/w1 and Ti/w2 and stop Ti/w3 (if it is running): 			
SIP Parameter values:	9,	<u> </u>		
ISUP Parameter values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
			←	404/484
			→	ACK
	SAM	→	→	INVITE
			←	404/484
			→	ACK
	SAM	→	→	INVITE
			←	404/484
			→	ACK
	SAM	→	→	INVITE
	ACM	← Ringing tone	←	180 Ringing
	ANM	←	← →	200 OK INVITE ACK
		Conv	ersation	
		→	→	BYE
	RLC		-	200 OK BYE

6.2.2.3 Sending of the ACM message

TP303001	SIP reference: RFC 32	61 [6]	ISUP reference:	
			ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4	
			and 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Sending o	f the ACM mess	sage	
SIP selection	PICS 3/1			
criteria:				
ISUP selection	PICS 4/9 AND NOT PICS 4/17			
criteria:				
Test purpose:	ACM is sent after the determina	ation of address	s complete indication in the SUT	
	Francisco de estate a CUT in India asset		and LANA are a second activities with a second at	
	called party number and the s		an IAM message containing the complete	
	called party number and the s	senaing comple	ete indication.	
	Sends the INVITE messag	e to called user	and starts Ti/w2	
			essage with the CPS indicator set to "no	
			ory indicator set to "no indication(00)" or	
			0)", the interworking indicator set to	
			ndicator set to "ISUP not used all the way", the	
	ISDN access indicator se			
SIP Parameter		<u> </u>		
values:				
ISUP Parameter	IAM; Called party number: cor	mplete number		
values:	ACM, CPS indicator: no indica			
	Called party's category indica	ator: no indication	on(00) or ordinary subscriber (01) or payphone	
	(10)			
	interworking indicator: interworking encountered (1)			
	ISUP indicator: ISUP not used			
	ISDN access indicator: "termin			
Comments:	ISUP/BICC	SUT	SIP	
	IAM →		→ INVITE	
		/w2 expired		
	ACM(no indication) ←			
	CPG ←	•	← 180 Ringing	
		•	ng tone	
	ANM ←		← 200 OK INVITE	
		•	→ ACK	
		Conve	rsation	
	→	•	→ BYE	
	RLC ←		← 200 OK BYE	

TP303002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4
		and 7.2.3.2.5
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM mes	ssage
SIP selection	PICS 2/3 AND PICS 3/1	
criteria:		
ISUP selection criteria:	PICS 4/9 AND PICS 4/17	
Test purpose:	64 kBit/s unrestricted call, ACM is sent after	the determination of address complete
	indication in the SUT	,
	Ensure that the SUT in Idle state, on receipt o	
	called party number and the sending comp	lete indication:
	 Sends the INVITE message to called use 	r and starts Ti/w2
		essage with the CPS indicator set to "no
	indication (00)", the Called party's categ	ory indicator set to "no indication(00)" or
		10)", the interworking indicator set to "no
		ndicator set to "ISUP used all the way", the
SIP Parameter	ISDN access indicator set to "terminatin INVITE: SDP a=rtpmap: <dynamic-pt> CLEAI</dynamic-pt>	
values:	INVITE: SDP a=ripmap. <uyilamic-pt> CLEAR</uyilamic-pt>	RIVIODE/6000
ISUP Parameter	IAM; Called party number: complete number	, TMR: "64 kbit/s unrestricted"
values:	ACM, CPS indicator: no indication (00)	
		ion(00) or ordinary subscriber (01) or payphone
	(10)	1 (0)
	interworking indicator: no interworking enco ISUP indicator: ISUP used all the way	untered (0)
	ISDN access indicator: "terminating access IS	DN"
Comments:	ISUP/BICC SUT	SIP
	IAM →	→ INVITE
	Ti/w2 expired	
	ACM(no indication) ←	
	CPG ←	← 180 Ringing
		ng tone
	ANM ←	← 200 OK INVITE
		→ ACK
		ersation → BYF
	RLC ←	→ BYE 200 OK BYE
	INLO T	200 ON DIE

TP303003	SIP reference: RFC 3261 [6		ISUP reference: 83 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5
TSS reference:	ISUP-SIP /Basic call/ Sending of the	ACM message	
SIP selection	PICS 3/1		
criteria:			
ISUP selection	PICS 4/9 AND NOT PICS 4/17		
criteria:	1011		
Test purpose:	ACM is sent after the maximum nur received Ensure that the SUT in Idle state, on number of digits used in the natio	receipt of an IA	M message containing the maximum
	indication (00)", the Called part "ordinary subscriber (01)" or "pa	e ACM messag y's category in yphone (10)", th the ISUP indica	e with the CPS indicator set to "no dicator set to "no indication(00)" or the interworking indicator set to tor set to "ISUP not used all the way", the
SIP Parameter			
values:			
ISUP Parameter	IAM; Called party number: complet	e number	
values:	ACM, CPS indicator: no indication (or ordinary subscriber (01) or payphone
	(10)	no maication(oo	of ordinary subscriber (01) or payprione
	interworking indicator: interworking	g encountered (1)
	ISUP indicator: ISUP not used all th	ie way	
	ISDN access indicator: "terminating		
Comments:	ISUP/BICC	SUT	SIP
	IAM →		→ INVITE
		2 expired	
	ACM(no indication) ← CPG ←		400 Dinning
	CPG •		← 180 Ringing
	ANM ←	Ringing ton	€ 200 OK INVITE
	, 4444		→ ACK
		Conversation	11211
	→		→ BYE
	RLC ←		€ 200 OK BYE

TP303004	SIP reference: RFC 3261 [6]	ES 283	ISUP reference: 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message			
SIP selection	PICS 2/3 AND PICS 3/1			
criteria:				
ISUP selection	PICS 4/9 AND PICS 4/17			
criteria:				
Test purpose:	64 kBit/s call, ACM is sent after the maxinumbering plan received Ensure that the SUT in Idle state, on receinumber of digits used in the national n	ipt of an IAM	message containing the maximum	
	 Sends the INVITE message to the called user and starts Ti/w2. When Ti/w2 is expired, sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered", the ISUP indicator set to "ISUP used all the way", the ISDN access indicator set to "terminating access ISDN". 			
SIP Parameter	INVITE: SDP a=rtpmap: <dynamic-pt> CL</dynamic-pt>	_EARMODE/8	3000	
values:				
ISUP Parameter	IAM; Called party number: complete num	nber		
values:	ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: no interworking encountered ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access ISDN"			
Comments:	ISUP/BICC SUT		SIP	
	IAM → Ti/w2 exp	→ pired	INVITE	
	ACM(no indication) ←			
	CPG ←	←	180 Ringing	
	F	Ringing tone		
	ANM ←	←	200 OK INVITE	
			ACK	
	_	Conversation		
	→	→	BYE	
	RLC ←		200 OK BYE	

TP303005	SIP reference: RFC 3261 [6]		ISUP reference:	
		ES 283	027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4	
			and 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 3/1			
criteria:				
ISUP selection	PICS 4/9 AND NOT PICS 4/17			
criteria:				
Test purpose:	ACM is sent after sufficient number of digital party. Ensure that the SUT in Idle state, on receipt called party number where the end of addre	f an IAM r	nessage containing the complete	
	called party number to indicate that a sufficie route the call to the called party:	nt numbe	r of digits has been received to	
	 Sends the INVITE message to the called user and starts Ti/w2. When Ti/w2 is expired, sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 			
SIP Parameter	IODIT decess indicator set to terminatin	g access	HOIT IODIV .	
values:				
ISUP Parameter	IAM; Called party number: complete number			
values:	ACM, CPS indicator: no indication (00)			
	Called party's category indicator: no indica	tion(00) or	ordinary subscriber (01) or payphone	
	(10)			
	interworking indicator: interworking encount	ered (1)		
	ISUP indicator: ISUP not used all the way			
	ISDN access indicator: "terminating access	non-ISDN		
Comments:	ISUP/BICC SUT	_	SIP	
	IAM →	. →	INVITE	
	Ti/w2 expired	I		
	ACM(no indication)	_	100 D: :	
	CPG ←	+	180 Ringing	
	_	ing tone	000 OK INN/ITE	
	ANM ←	(200 OK INVITE	
		→	ACK	
		ersation	DVE	
	→	→	BYE	
	RLC ←	+	200 OK BYE	

TP303006	SIP reference: RFC 3261 [6]	ISUP reference:		
		ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4		
		and 7.2.3.2.5		
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 2/3 AND PICS 3/1			
criteria:				
ISUP selection	PICS 4/9 AND PICS 4/17			
criteria:	241511			
Test purpose:		sufficient number of digits has been received		
	to route the call to the called party			
	Ensure that the SUT in Idle state, on receipt of	an IAM message containing the complete		
	called party number where the end of address			
	called party number to indicate that a sufficie			
	route the call to the called party:			
	 Sends the INVITE message to called use 			
	 When Ti/w2 is expired, sends the ACM m 			
	indication (00)", the Called party's categ			
		0)", the interworking indicator set to "no		
		ndicator set to "ISUP used all the way", the		
OID Davis at a s	ISDN access indicator set to "terminatin			
SIP Parameter	INVITE: SDP a=rtpmap: <dynamic-pt> CLEAF</dynamic-pt>	RMODE/8000		
values:	IAM; Called party number: complete number	TMD: "64 kbit/o uprostricted"		
values:	ACM, CPS indicator: no indication (00)	, TWR. 64 KDIVS unlestricted		
values.		ion(00) or ordinary subscriber (01) or payphone		
	(10)	ion(ob) or oraniary dabberiber (on) or payprione		
	interworking indicator: no interworking enco	untered (0)		
	ISUP indicator: ISUP used all the way	· ,		
	ISDN access indicator: "terminating access I			
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	Ti/w2 expired			
	ACM ←			
	CPG ←	180 Ringing		
	Ringing tone			
	ANM ←	← 200 OK INVITE		
		→ ACK		
		ersation		
	→	→ BYE		
	RLC ←	← 200 OK BYE		

TP303007	SIP reference: RFC 3261 [6]	ES 202 (ISUP reference:	
		ES 203 (027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 3/1			
criteria:				
ISUP selection	NOT PICS 4/17			
criteria:				
Test purpose:	ACM is sent determined by the expiration time	^{r /} I/W1		
	Ensure that the SUT in Idle state, on receipt of called party number where the end of address timer T _{I/W1} after the receipt of the latest ad	s signallir	ng is determined by the expiration	
	 sends the INVITE message to the called user. Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 			
SIP Parameter	-			
values:				
ISUP Parameter	IAM; Called party number: complete number			
values:	ACM, CPS indicator: no indication (00) Called party's category indicator: no indicat	ion(00) or	ordinary subscriber (01) or navabane	
	(10)	1011(00) 01	ordinary subscriber (01) or payprione	
	interworking indicator: interworking encountered (1)			
	ISUP indicator: ISUP not used all the way	()		
	ISDN access indicator: "terminating access r	non-ISDN"		
Comments:	ISUP/BICC SUT		SIP	
	IAM → Start T _{I/W1}			
	T _{I/W1} expiry			
	ACM(no indication) ←	→	INVITE	
	CPG ←	(180 Ringing	
	Ringing tone	_		
	ANM ←	(200 OK INVITE	
	0	→	ACK	
		ersation	DVE	
	→ RLC ←	→ ←	BYE 200 OK BYE	
	INLO T		ZUU UN DIE	

TP303008	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message		
SIP selection	PICS 2/3 AND PICS 3/1		
criteria:			
ISUP selection	PICS 4/17		
criteria:	64 k Dit/o unreatriated call ACM is contactor of	latorminad by the expiration timer T	
Test purpose:	64 kBit/s unrestricted call, ACM is sent after of	etermined by the expiration timer 1/1/W1	
	containing the complete called party number	of an IAM message, TMR=64 kBit/s unrestricted by where the end of address signalling is signer the receipt of the latest address message:	
	 Sends the INVITE message to called user. Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered (0)", the ISUP indicator set to "ISUP used all the way", the ISDN access indicator set to "terminating access ISDN". 		
SIP Parameter	INVITE: SDP a=rtpmap: <dynamic-pt> CLEA</dynamic-pt>	RMODE/8000	
values:			
ISUP Parameter	IAM; Called party number: complete number	r, TMR: "64 kbit/s unrestricted"	
values:	ACM, CPS indicator: no indication (00)	tion(00) or ordinary subscriber (01) or payphone	
	(10)	tion(ob) of ordinary subscriber (or) or payprione	
	interworking indicator: no interworking enco	ountered (0)	
	ISUP indicator: ISUP used all the way	` ,	
	ISDN access indicator: "terminating access		
Comments:	ISUP/BICC SUT	SIP	
	→ Start T _{I/W1}		
	T _{I/W1} expiry		
	ACM(no indication)	→ INVITE	
	CPG ←	← 180 Ringing	
	Ringing tone ANM	€ 200 OK INVITE	
	AINIVI	→ ACK	
	Conv	rersation	
	→	→ BYE	
	RLC +	€ 200 OK BYE	
		200011212	

TP303009	SIP reference:			ISUP reference: 283 027 [1], clauses 7.2.3.2.5 and 7.2.3.2.1.4	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message				
SIP selection criteria:	PICS 3/2				
ISUP selection criteria:	NOT PICS 4/17				
Test purpose:	ACM is sent determined	d by the expiration time	er T _{I/W2}		
	an IAM message contain has been received (star procedure):	ning the minimum nu t timer TI/W2 and invo	mber of c ke the app	toward the SIP network, on receipt of digits required for routing the call propriate outgoing SIP signalling	
				after the expiration of T _{I/W2.}	
	party's category in "payphone (10)", the	ndicator set to "no ind ne interworking indica to "ISUP not used all	ication(00 ator set to	set to "no indication (00)", the Called b)" or "ordinary subscriber (01)" or "interworking encountered (1)", the the ISDN access indicator set to	
SIP Parameter					
values:					
ISUP Parameter	IAM; Called party num				
values:	ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone				
	(10)				
	interworking indicator: interworking encountered (1)				
	ISUP indicator: ISUP not used all the way				
	ISDN access indicator		non-ISDN		
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→ Start T _{I/W1}			
	SAM	→ Start T _{I/W1}			
	SAM	→ Start T _{I/W2}	→	INVITE	
		T _{I/W2} expiry			
	ACM(no indication)	←			
	CPG	←	←	180 Ringing	
		-	ing tone		
	ANM	←	(200 OK INVITE	
		^	→	ACK	
			ersation	DVE	
	DI C	→	→	BYE	
	RLC			200 OK BYE	

TP303010	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1.4 and 7.2.3.2.5		
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 2/3 AND PICS 3/2			
criteria:				
ISUP selection	PICS 4/17			
criteria:				
Test purpose:	64 kBit/s unrestricted call, ACM is sent after de	determined by the expiration timer T _{I/W2}		
	Ensure that the SUT if overlap addressing is to an IAM message, TMR=64 kBit/s unrestricted required for routing the call has been recei appropriate outgoing SIP signalling procedure • Sends an INVITE message to the called units of the called uni	ived (start timer TOIW2 and invoke the e):		
	party's category indicator set to "no ind "payphone (10)", the interworking indicator ISUP indicator set to "ISUP used all the "terminating access ISDN".	•		
SIP Parameter	INVITE: SDP a=rtpmap: <dynamic-pt> CLEAF</dynamic-pt>	RMODE/8000		
values:				
ISUP Parameter	IAM; Called party number: complete number	r, TMR: "64 kbit/s unrestricted"		
values:	ACM, CPS indicator: no indication (00)			
	Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone			
	(10) interworking indicator: no interworking encountered (0)			
	ISUP indicator: ISUP used all the way			
	ISDN access indicator: "terminating access I	ISDN"		
Comments:	ISUP/BICC SUT	SIP		
	IAM → Start T _{I/W1}			
	SAM → Start T _{I/W1}			
	SAM → Start T _{I/W2}	→ INVITE		
	T _{/IW2} expiry			
	ACM(no indication) ←			
	CPG ←	← 180 Ringing		
	Ringing tone	• •		
	ANM ←	← 200 OK INVITE		
		→ ACK		
		versation versation		
	→	→ BYE		
	RLC ←	€ 200 OK BYE		

TP303011	SIP reference: RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.2.5		
TSS reference:					
SIP selection	ISUP-SIP /Basic call/Sending of the ACM message PICS 3/1				
criteria:	F1C3 3/1				
ISUP selection	NOT PICS 4/9 AND NOT PICS 4/17				
criteria:	1101 1 100 4/3 / 110 NOT 1 100 4/17				
Test purpose:	ACM is sent after 180 Ringing was received	d			
	 Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, on receipt of a 180 Ringing message: Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to 				
SIP Parameter	"terminating access non-ISDN".				
values:					
ISUP Parameter	IAM; Called party number: complete number	per			
values:	ACM, CPS indicator: subscriber free (01)				
	Called party's category indicator: no indi	cation(00) o	r ordinary subscriber (01) or payphone		
	(10)				
	interworking indicator: interworking enco	untered (1)			
	ISUP indicator: ISUP not used all the way	o non ISDN			
Comments:	ISDN access indicator: "terminating access ISUP/BICC SUT	92 HOH-19DI	SIP		
Comments.	IAM →	→	INVITE		
	ACM ←	-	180 Ringing		
	Ringing tone	•	100 Kinging		
	ANM ←	←	200 OK INVITE		
		→	ACK		
	Conversation	-	7.0.0		
	→	→	BYE		
	RLC ←	É	200 OK BYE		
	<u> </u>				

TP303012	SIP reference: RFC 3261 [6]	ES	ISUP reference: 3 283 027 [1], clause 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection criteria:	PICS 2/3 AND PICS 3/1			
ISUP selection criteria:	NOT PICS 4/9 AND PICS 4/17			
Test purpose:	64 kBit/s unrestricted call, ACM is sent after 1	80 Ringing	was received	
	Ensure that the SUT in Idle state, on receipt of containing the complete called party number			
	• Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered (0)", the ISUP indicator set to "ISUP used all the way", the ISDN access indicator set to "terminating access ISDN".			
SIP Parameter values:	INVITE: SDP a=rtpmap: <dynamic-pt> CLEAF</dynamic-pt>	RMODE/80	00	
ISUP Parameter	IAM; Called party number: complete number	TMR: "64	kbit/s unrestricted"	
values:	ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)			
	interworking indicator: no interworking enco ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access I			
Comments:	ISUP/BICC SUT		SIP	
	IAM →	→	INVITE	
	ACM ←	←	180 Ringing	
	Ringing tone			
	ANM ←	←	200 OK INVITE	
	Conversation	→	ACK	
	→	→	BYE	
	RLC ←	-	200 OK BYE	

TP303013	SIP reference: RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.2.4		
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message				
SIP selection	PICS 4/15				
criteria:					
ISUP selection	NOT PICS 4/9 AND NOT PICS 4/17				
criteria:					
Test purpose:	The SUT supports the P-Early-Media header				
SIP Parameter values:	Ensure that the SUT, on receipt of an IAM message containing the complete called party number, where the O-MGCF is supporting the P-Early-Media header as a network option, on the reception of the first 180 Ringing that includes a P-Early-Media header authorizing early media, sends the ACM message with the CPS indicator set to "no indication (00)", Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", OBCI "in -band information" set to: yes. 180 Ringing: P-Early-Media header				
ISUP Parameter	IAM; Called party number: complete number				
values:	ACM; CPS indicator: no indication (00),				
Comments:	OBCI: in -band information: yes ISUP/BICC SUT		SIP		
Comments.	IAM →	→	INVITE		
	IAW	•	IIIVII L		
	ACM ←	←	180 Ringing		
	Ringing tone	•	100 Kinging		
	ANM	+	200 OK INVITE		
		÷	ACK		
	Conv	ersation			
	→	→	BYE		
	RLC ←	<u> </u>	200 OK BYE		

TP303014	SIP reference: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message			
SIP selection	PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	180 received, mapping of PSTN XML	Progresslindicato	r #7 into the ACM BCI	
		th Progresslindic the Called Party	ator # 7 (Terminating user ISDN) "s Status (CPS) indicator set to "	
	subscriber free (01)", the Called party"s category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered (0)", the ISUP indicator set to "ISUP used all the way" and the ISDN access indicator set to "ISDN" and if included the access delivery information is set to "Set-up message generated".			
SIP Parameter	180 Ringing;			
values:	PSTN XML body with ProgressIndicate	or # 7 (Terminatin	g user ISDN)	
ISUP Parameter	ACM, CPS indicator: subscriber free (01)			
values:		o indication(00) o	r ordinary subscriber (01) or payphone	
	(10)			
	interworking indicator: no interworking encountered (0)			
	ISUP indicator: ISUP used all the way			
	ISDN access indicator: terminating access isISDN access delivery information: Set-up message generated (IF PRESENT)			
Comments:		UT	SIP	
Comments.	IAM →	01	511	
	I/AIVI	→	INVITE	
	ACM ←	É	180 Ringing	
	Ringing tone	`	100 Kinging	
	ANM	←	200 OK INVITE	
	7.000	÷	ACK	
		Conversation	,,,,,,	
	→	Conversation	BYE	
	RLC C	-	200 OK BYE	
	INLO T		200 OR DIL	

TP303015	SIP refer	rence: RFC 3261 [6]	-	ISUP reference: S 283 027 [1], clause 7.2.3.2.5	
TSS reference:					
		ISUP-SIP /Basic call/ Sending of the ACM message			
SIP selection criteria:	PICS 4/18				
ISUP selection					
criteria:					
Test purpose:	180 received m	anning of PSTN YMI Progress	sIndicato	r into Progress Indicator contained in	
rest purpose.	the ATP in the se	ent ACM		·	
	Ensure that the SUT, if an ACM has not been already sent, on receipt the 180 Ringing message, with the PSTN XML body with Progress indicator # 7 (Terminating user ISDN) and a PSTN XML ProgressIndicator set to PI_VALUE. The ATP does not contain the ProgressIndicator #7.				
	sends th	e ACM message with the CP		or set to " subscriber free (01)" and the g the progress indicator PI_VALUE.	
SIP Parameter values:	180 Ringing; PSTN XML ProgressIndicator : PI_VALUE (PIXIT)				
ISUP Parameter	ACM, CPS indicator: subscriber free (01)				
values:	ATP progress inc	dicator: PI_VALUE (PIXÍT)			
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→			
			→	INVITE	
	ACM	←	←	180 Ringing	
		Ringing tone			
	ANM	←	←	200 OK INVITE	
			→	ACK	
		Conv	ersation		
		→	→	BYE	
	RLC	←	←	200 OK BYE	

Values and additional selection criteria for test purposes TP303015				
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)			
VA 02	PI VALUE = Destination address is non-ISDN (#2)			

TP303016	SIP reference: RI	FC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.2.5
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
		<u> </u>	sage	
SIP selection criteria:	PICS 3/1 AND NOT PICS	0 4/15		
ISUP selection criteria:	NOT PICS 4/9 AND NOT	PICS 4/17		
Test purpose: SIP Parameter values:	P-Early-Media header not supported, 183 is not interworked sending complete indication received Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication, on receipt of a 183 Session Progress: • Sends the INVITE message to called user. • No ISUP message is sent backward.			
ISUP Parameter values:	IAM; Called party number	er: complete number		
Comments:	ISUP/BICC	SUT		SIP
	IAM -	>	→	INVITE
			←	183 Session Progress
	ACM	÷	←	180 Ringing
		Ringi	ng tone	
	ANM	•	←	200 OK INVITE
			→	ACK
		Conve	ersation	
	_	•	→	BYE
	RLC •	-	←	200 OK BYE

TP303017	SIP reference	e: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.5
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 4/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	P-Early-Media header supported, 183 is interworked, an ACM no indication is sent Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, where the O-MGCF is supporting the P-Early-Media header if the 183 Session Progress contains a P-Early_Media header authorizing early media • sends the ACM message with the CPS indicator set to " no indication (00)", the Called party"s category indicator set to "no indication(00)", OBCI "in -band information" set to:			
SIP Parameter	yes. 183 Session Progres	s that includes a P-Early-	Media he	ader authorizing early media
values:		,		3
ISUP Parameter	IAM; Called party number: complete number			
values:	ACM; CPS indicator			
	OBCI: in -band inform			
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM(no indication)	←	←	183 Session Progress
	CPG(Alerting)	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
		→	→	BYE
	RLC	+	+	200 OK BYE

TP303018	SIP refere	ence: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 9.2.3.3.12
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message			
SIP selection criteria:	PICS 3/1			
ISUP selection criteria:	NOT PICS 4/15			
Test purpose:	P-Early-Media header not supported, 183 is not interworked maximum number of digits used in the national numbering plan received Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan on receipt of a 183 Session Progress: No ISUP message is sent backward.			
SIP Parameter values:				
ISUP Parameter values:	IAM; Called party number: complete number			
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
			←	183 Session Progress
	ACM	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
		→	→	BYE
	RLC		+	200 OK BYE

TP303019	SIP refere	nce: RFC 3261 [6]		ISUP reference:
			E:	S 283 027 [1], clause 9.2.3.3.12
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message			
SIP selection	PICS 3/1	-		
criteria:				
ISUP selection	NOT PICS 4/15			
criteria:				
Test purpose:	,	nder not supported, 183 is no oute the call to the called pa		orked sufficient number of digits has wed
	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party on receipt of a 183 Session Progress:			
SIP Parameter	NO BICC/ISUR	P message is sent backward	1.	
values:				
ISUP Parameter				
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
			←	183 Session Progress
	ACM	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
		→	→	BYE
	RLC	+	+	200 OK BYE

TP303020	SIP reference: RFC 3261 [6]	E	ISUP reference: 5 283 027 [1], clause 9.2.3.3.12
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message		
SIP selection	PICS 3/1 NOT PICS 4/15		
criteria:			
ISUP selection	NOT PICS 4/9		
criteria:			
Test purpose:	183 received after T _{I/W1} expired, P-Early-Me Ensure that the SUT in Idle state, on receipt c called party number where the end of addre	of an IAM r	nessage containing the complete
	timer T _{I/W1} after the receipt of the latest add		
	Progress:		9
	l Togress.		
	No ISUP message is sent backward.		
SIP Parameter			
values:			
ISUP Parameter			
values:			
Comments:	ISUP/BICC SUT		SIP
	IAM →		
	T _{I/W1} expiry		
	ACM(no indication) ←	→	INVITE
		←	183 Session Progress
	CPG(alerting) ←	←	180 Ringing
	Ringing tone		
	ANM ←	←	200 OK INVITE
		→	ACK
	Conv	ersation/	
	→	→	BYE
	RLC ←	<u>+</u>	200 OK BYE

TP303021	SIP reference: RFC 3261 [6]	ES	ISUP reference: S 283 027 [1], clause 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection criteria:	PICS 4/15 AND PICS 4/18			
ISUP selection				
criteria:				
Test purpose:	183 received, mapping of PSTN XML Progres	slindicator	#7 into the ACM BCI	
	Ensure that the SUT, on receipt of the 183 Se ProgressIndicator # 7 (Terminating user ISD	N)		
	 sends the ACM message with the CPS indicator set to "no indication (00)", the Called party"s category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered (0)", the ISUP indicator set to "ISUP used all the way" and the ISDN access indicator set to "ISDN" and if included the access delivery information is set to "Set-up message generated". 			
SIP Parameter	183 Session Progress; PSTN XML Progress	ndicator #	# 7 (Terminating user ISDN)	
values:				
ISUP Parameter	ACM, CPS indicator: no indication (00) Called party"s category indicator: no indication(00) or ordinary subscriber (01) or payphone			
values:	(10)			
	interworking indicator: no interworking enco	untered (0	1	
	ISUP indicator: ISUP used all the way			
	ISDN access indicator: ISDN			
	access delivery information: Set-up message	ge generat	ed (IF PRESENT)	
Comments:	ISUP/BICC SUT		SIP	
	IAM →	→	INVITE	
	ACM ←	←	183 Session Progress	
	CPG ←	←	180 Ringing	
	Ringing tone			
	ANM ←	←	200 OK INVITE	
		→	ACK	
	Conv	ersation		
	→	→	BYE	
	RLC ←	+	200 OK BYE	

TP03022	SIP refere	nce: RFC 3261 [6]		ISUP reference:
11 00022	On referen	1100: IXI O 0201 [0]	E	S 283 027 [1], clause 7.2.3.2.5
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 4/15 AND PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	183 received, mapping of PSTN XML ProgressIndicator into Progress Indicator contained in the ATP in the sent ACM			
	body with Progress	s indicator # 7 (Terminating	user ISDI	ogress message with the PSTN XML N) containing the PSTN XML asIndicator #7 is not interworked.
	 sends the ACM message with the CPS indicator set to "no indication (00)", the Called party"s category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered (0)", the ISUP indicator set to "ISUP used all the way", the ISDN access indicator set to "ISDN", the Access Transport Parameter (ATP) containing the progress indicator set to PI_VALUE and if included the access delivery information is set to "Set-up message generated". 			
SIP Parameter values:	183 Session Progr	ess; PSTN XML Progress	Indicator:	PI_VALUE
ISUP Parameter	183 Session Progress, CPS indicator: no indication (00)			
values:	Called party"s category indicator: no indication(00) or ordinary subscriber (01) or payphone			
	(10)			
	interworking indicator: no interworking encountered (0) ISUP indicator: ISUP used all the way			
	ISDN access indicator.			
	ATP progress ind			
		nformation: Set-up messag	ge genera	ted (IF PRESENT)
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM	←	←	183 Session Progress
	CPG	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
		→	→	BYE
	RLC		+	200 OK BYE

Values for test purposes TP303022			
VA_01	PI_VALUE: Call is not end-to-end ISDN: further call progress information is available in-		
	band (#1)		
VA_02	PI_VALUE: Destination address is non-ISDN (#2)		

TP303023	SIP re	ference: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.5
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	NOT PICS 4/15 A	ND NOT PICS 4/18		
criteria:				
ISUP selection				
criteria:				
Test purpose:	183 received, mapping of PSTN XML ProgressIndicator into Progress Indicator contained in the ATP and mapping of PSTN XML ProgressIndicator #7 BClis not supported Ensure that the SUT, on receipt of the 183 Session Progress message with the PSTN XML body with Progress indicator #7 (Terminating user ISDN) and containing the PSTN XML ProgressIndicator set to PI_VALUE, does not send the ACM message.			
SIP Parameter		ress; PSTN XML Progress	Indicator	PI VALUE
values:		, ,		
ISUP Parameter				
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
			←	183 Session Progress
	ACM	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
		→	→	BYE
	RLC		+	200 OK BYE

Values for test purposes TP303023			
VA_01	PI_VALUE: originating address is non-ISDN (#3)		
VA_02	PI_VALUE: Call has returned to ISDN (#4)		

TP303024	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.3.3.2.3	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM mes	ssage	
SIP selection criteria:	PICS 3/1 AND PICS 4/5 AND PICS 4/11		
ISUP selection criteria:	PICS 4/2 AND PICS 4/9 AND NOT PICS 4/17	•	
Test purpose:	Preconditions requested, ACM is sent after the in the SUT Ensure that the SUT in Idle state, on receipt of	e determination of address complete indication f an IAM message containing the complete	
	called party number, the sending complete on this circuit (ISUP) or COT is expected (BIC	indication, and the continuity check is required indication.	
	received.	riand starts 17/w2. iil a successful continuity indication has been ndicator set to "no indication (00)", the Called	
	party's category indicator set to "no inc "payphone (10)", the interworking indicator	lication(00)" or "ordinary subscriber (01)" or ator set to "interworking encountered (1)", the the way", the ISDN access indicator set to	
SIP Parameter values:			
ISUP Parameter	IAM; Called party number: complete numbe	r	
values:	ACM, CPS indicator: no indication (00)		
	Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone		
	(10)		
	interworking indicator: interworking encount	tered (1)	
	ISUP indicator: ISUP used all the way		
	ISDN access indicator: "terminating access		
Comments:	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
		← 183 Session Progress	
		→ PRACK	
		← 200 OK PRACK	
	COT →	→ UPDATE	
		← 200 OK UPDATE	
	ACM ← Ti/w2 expired		
	CPG(Alerting)	← 180 Ringing	
		→ PRACK	
		← 200 OK PRACK	
	Ringing tone		
	ANM ←	← 200 OK INVITE	
		→ ACK	
		rersation	
	→	→ BYE	
	RLC ←	← 200 OK BYE	

TP303025	SIP reference:	RFC 3261 [6]		ISUP reference:
			Е	S 283 027 [1], clause 7.3.3.2.3
TSS reference:		ending of the ACM mes	sage	
SIP selection criteria:	PICS 3/1 AND PICS 4/	5 AND PICS 4/11		
ISUP selection criteria:	PICS 4/2 AND PICS 4/	9 AND PICS 4/17		
Test purpose:	64 kBit/s, Precondition indication in the SUT	s requested, ACM is se	nt after th	ne determination of address complete
	called party number,		indication	message containing the complete n and the continuity check is required
	 The SUT shall with received. 	· ·	l a succe	ssful continuity indication has been
	party's category i "payphone (10)", t ISUP indicator se	indicator set to "no ind he interworking indica et to "ISUP used all the	ication(00 I tor set to	set to "subscriber free (01)", the Called 0)" or "ordinary subscriber (01)" or 0 "no interworking encountered (0)", the ISDN access indicator set to
OID Danson stars	"terminating acces		NAODE (2000
SIP Parameter values:	INVITE: SDP a=rtpmap	•		
ISUP Parameter	IAM; Called party nun		, TMR: "6	64 kbit/s unrestricted"
values:	ACM, CPS indicator:		(0.0)	" (04)
		ry indicator: no indicat	ion(00) o	r ordinary subscriber (01) or payphone
	(10) interworking indicato	r: no interworking enco	untered (0)
	ISUP indicator: ISUP		untereu (0)
	ISDN access indicato		SDN"	
Comments:	ISUP/BICC	SUT	_	SIP
	IAM	→	→	INVITE
			←	183 Session Progress
			→	PRACK
			+	200 OK PRACK
	СОТ	→	→	UPDATE
			-	200 OK UPDATE
	ACM(no indication)	← Ti/w2 expired	_	200 01(01 2/112
	CPG	←	←	180 Ringing
	01 0	•	÷	PRACK
			÷	200 OK PRACK
			•	200 OKT KAOK
	ANM	←	←	200 OK INVITE
		=	÷	ACK
		Conv	ersation	,
		→	••••••••••••••••••••••••••••••••••••••	BYE
	RLC	+	ŕ	200 OK BYE
	INLO			ZUU UN DTE

TP303028	SIP referen	ce: RFC 3261 [6]		ISUP reference:		
				S 283 027 [1], clause 7.3.3.2.3		
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message					
SIP selection criteria:	PICS 3/2 AND PICS 4/5 AND PICS 4/11					
ISUP selection criteria:	PICS 4/2 AND NOT	PICS 4/17				
Test purpose:	Preconditions reque	ested, ACM is sent after ex	kpiration o	f timer T _{I/W2}		
	Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received, start timer TI/W2 and invoke the appropriate outgoing SIP signalling procedure and the continuity check is required on this circuit (ISUP):					
	The SUT shall veceived.	withhold sending ACM unt	il a succe	ssful continuity indication has been		
	 Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non ISDN". 					
SIP Parameter values:		nap: <dynamic-pt> CLEA</dynamic-pt>	RMODE/8	3000		
ISUP Parameter	IAM: Called party n	umber: complete numbe	r			
values:	ACM, CPS indicato		•			
			tion(00) o	r ordinary subscriber (01) or payphone		
	(10)		()	, , , , , ,		
		ator: interworking encoun	tered (1)			
	ISUP indicator: ISU	JP not used all the way				
	ISDN access indica	ator: "terminating access	non ISDN	"		
Comments:	ISUP/BICC	SUT		SIP		
	IAM	→	→	INVITE		
			(183 Session Progress		
			→	PRACK		
			←	200 OK PRACK		
	COT	→	→	UPDATE		
			←	200 OK UPDATE		
		T _{I/W2} expiry				
	"					
	ACM(no indication)	(_	190 Binging		
	CPG(Alerting)	~	-	180 Ringing		
			→	PRACK		
		Dinaina tan -	+	200 OK PRACK		
	A N I N 4	Ringing tone	-	OOO OK INIVITE		
	ANM	←	(200 OK INVITE		
			→	ACK		
		_	ersation			
		→	→	BYE		
	RLC		+	200 OK BYE		

TP303029	SIP refe	erence: RFC 3261 [6]		ISUP reference:			
TSS reference:	ICLID CID /Dooi	is sall/Canding of the INIVITE		S 283 027 [1], clause 7.3.3.2.3			
SIP selection	ISUP-SIP /Basic call/Sending of the INVITE message						
criteria:	PICS 2/3 AND PICS 3/2 AND PICS 4/5 AND PICS 4/11						
ISUP selection criteria:	PICS 4/2 AND						
Test purpose:	64 kBit/s call, F	Preconditions requested, ACM	is sent aft	er expiration of timer T _{I/W2}			
	an IAM messag has been receiv	ge containing the minimum n	umber of o	I toward the SIP network, on receipt of digits required for routing the call propriate outgoing SIP signalling ircuit (ISUP):			
	 The SUT s received. 	hall withhold sending ACM ur	ntil a succe	ssful continuity indication has been			
	 Sends the party's car payphone ISUP indic 	tegory indicator set to "no in (10)", the interworking indic	dication(00 cator set to	set to "subscriber free (01)", the Called 0)" or "ordinary subscriber (01)" or 0 "no interworking encountered (0)", the ISDN access indicator set to			
SIP Parameter values:	INVITE: SDP a	=rtpmap: <dynamic-pt> CLE/</dynamic-pt>	ARMODE/8	3000			
ISUP Parameter	IAM: Called pa	rty number: complete number	er. TMR: "6	34 kbit/s unrestricted"			
values:		icator: no indication (00)	,				
			ation(00) o	r ordinary subscriber (01) or payphone			
	(10)		, ,	, , ,			
		ndicator: no interworking end	ountered (0)			
		r: ISUP used all the way					
		ndicator: "terminating access	ISDN"				
Comments:	ISUP/BICC	SUT		SIP			
	IAM	→	→	INVITE			
			←	183 Session Progress			
			→	PRACK			
			←	200 OK PRACK			
	COT	→	→	UPDATE			
			←	200 OK UPDATE			
	T _{I/W2} expiry						
	ACM	←					
	CPG	`	_	180 Ringing			
		•	→	PRACK			
			-	200 OK PRACK			
		Dinging tons	~	ZOU ON FRAGR			
	ANIM	Ringing tone	4 -	200 OK INIVITE			
	ANM	((200 OK INVITE			
		_	→	ACK			
		_	versation				
		→	→	BYE			
	RLC	←	<u> </u>	200 OK BYE			

TP303030	SIP reference: RFC 3261 [6]		ISUP reference:			
			S 283 027 [1], clause 7.3.3.2.3			
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message					
SIP selection criteria:	PICS 4/5 AND PICS 4/11					
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9 AND NOT PICS	4/17				
Test purpose:	180 received after precontitions met, an ACM	is sent				
	Ensure that the SUT in Idle state, on receipt of called party number, the continuity check is rof a 180 Ringing message:					
	Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN".					
SIP Parameter						
values:						
ISUP Parameter values:	IAM; Called party number: complete number ACM, CPS indicator: "subscriber free (01)" Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access non-ISDN"					
Comments:	ISUP/BICC SUT		SIP			
	IAM →	→	INVITE			
		←	183 Session Progress			
		→	PRACK			
		←	200 OK PRACK			
	COT →	→	UPDATE			
		←	200 OK UPDATE			
	ACM ←	←	180 Ringing			
		→	PRACK			
		←	200 OK PRACK			
	Ringing tone					
	ANM ←	-	200 OK INVITE			
	_	→	ACK			
		ersation	5)/5			
	→	→	BYE			
	RLC ←	<u>+</u>	200 OK BYE			

TP303031	SIP reference: RFC 3261 [6]		ISUP reference:				
			S 283 027 [1], clause 7.3.3.2.3				
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message						
SIP selection criteria:	PICS 2/3 AND PICS 4/5 AND PICS 4/11						
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9 AND PICS 4/17	,					
Test purpose:	64 kBit/s call, 180 received after precontitions	met, an A	CM is sent				
	Ensure that the SUT in Idle state, on receipt of called party number, the continuity check is expected (BICC) indication on receipt of a 180	required o	n this circuit (ISUP) or COT is				
	(01)" or "payphone (10)", the interwo encountered (0)", the ISUP indicato access indicator set to "terminating	et to "no in orking ind r set to "IS access IS	ndication(00)" or "ordinary subscriber icator set to "no interworking UP used all the way", the ISDN DN".				
SIP Parameter values:	INVITE: SDP a=rtpmap: <dynamic-pt> CLEA</dynamic-pt>	RMODE/8	000				
ISUP Parameter values:	IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication (10) interworking indicator: no interworking encountry indicator: ISUP used all the way ISDN access indicator: "terminating access	tion(00) or ountered (0	ordinary subscriber (01) or payphone				
Comments:	ISUP/BICC SUT		SIP				
	IAM →	→	INVITE				
		+	183 Session Progress				
		→	PRACK				
		+	200 OK PRACK				
	COT →	÷	UPDATE				
		+	200 OK UPDATE				
	ACM ←	÷	180 Ringing				
	/ tolvi	÷	PRACK				
		É	200 OK PRACK				
	Ringing tone	•	200 OKT KAOK				
	ANM ←	←	200 OK INVITE				
	7 (1414)	→	ACK				
	Conv	ersation	AOR				
	→	ersation →	BYE				
	RLC +	-	200 OK BYE				
	INLO T		ZOO ON DIL				

TP303032	SIP reference: RFC 3261 [6]	_	ISUP reference:			
			S 283 027 [1], clause 7.2.3.2.5			
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM mes	ssage				
SIP selection criteria:	PICS 2/3 AND PICS 4/18 AND PICS 4/19					
ISUP selection						
criteria:						
Test purpose:	Mapping of PSTN XML BearerCapability elem parameter sent in the ACM	ent contai	ined in the 180 into the TMU			
	Ensure on receipt of a 180 Ringing containes XML BearerCapability BC_VALUE, an ACM is BC_VALUE.					
	The BCI is set to: ISUP indicator: ISUP is used all the way					
	ISDN access indicator: ISDN					
	Interwoking indicator: Interworking not enount	ered				
SIP Parameter	INVITE;					
values:	PSTN XML first Bearer Capability: INVITE _					
	PSTN XML second Bearer Capability: INVIT	TE _BC2				
	180 Ringing PSTN XML BC: BC_VALUE and	XML PI #7	7			
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law					
values:	USI prime: Unrestr. Digital info T/A, G.711 A-I	aw				
	TMR: 64 kbit/s preferred					
	TMR prime: Speech/audio3Kbit/s					
	ACM: ISUP indicator: ISUP is used all the wa	У				
	ISDN access indicator: ISDN Interwoking indicator: Interworking not enount	orod				
	TMU: BC VALUE	cieu				
Comments:	ISUP/BICC SUT		SIP			
	IAM →	→	INVITE			
	ACM ←	-	180 Ringing			
	Ringing tone		3 3			
	ANM ←	←	200 OK INVITE			
		→	ACK			
	Conv	ersation				
	→	→	BYE			
	RLC (+	200 OK BYE			

TP303033	SIP reference: RFC 3261 [6]	F	ISUP reference: ES 283 027 [1], clause 7.2.3.2.5				
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message						
SIP selection	PICS 2/3 AND PICS 4/18 AND PICS 4/19	ssaye					
criteria:	11 100 2/3 AND 1 100 4/10 AND 1 100 4/13						
ISUP selection							
criteria:							
Test purpose:	Mapping of PSTN XML PearerCapability eler	nent conta	nined in the 183 into the TMU				
	parameter sent in the ACM						
	Ensure on receipt of a 183 Session Progress containes PSTN XML ProgressIndicator #7 and PSTN XML BearerCapability BC_VALUE, an ACM is sent containing the TMU Parameter BC_VALUE . The BCI is set to: ISUP indicator: ISUP is used all the way						
	ISDN access indicator: ISDN Interwoking indicator: Interworking not enoun	torod					
SIP Parameter	INVITE;	lereu					
values:	PSTN XML first Bearer Capability: INVITE	BC1					
	PSTN XML second Bearer Capability: INVI 183 Session Progress PSTN XML BC: BC_V	TE _BC2	I XML PI #7				
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law						
values:	USI prime: Unrestr. Digital info T/A, G.711 A-	law					
	TMR: 64 kbit/s preferred						
	TMR prime: Speech/audio3Kbit/s						
	ACM: ISUP indicator: ISUP is used all the way						
	ISDN access indicator: ISDN	ıy					
	Interwoking indicator: Interworking not enoun	tered					
	TMU: BC_VALUE						
Comments:	ISUP/BICC SUT		SIP				
	IAM →	→	INVITE				
	ACM(no indication)	←	100 00000000000000000000000000000000000				
	ACM ←	+	180 Ringing				
	Ringing tone	_					
	ANM ←	(200 OK INVITE				
	_	. →	ACK				
	_	ersation	DVE				
	→	→	BYE				
	RLC ←		200 OK BYE				

Values and selection criteria for test purpose TP303032a and TP303033				
Test purposes	ACM Parameter values	18x Provisional response values:	INVITE parameter value	
VA_01	TMU_VALUE: speech	PSTN XML: BC_VALUE: speech	PSTN XML INVITE BC1: speech INVITE BC2: unrestricted digital information with tones and announcements	
VA_02	TMU_VALUE: 3,1 kHz	PSTN XML: BC_VALUE: 3,1 kHz audio	PSTN XML INVITE BC1: 3,1 kHz audio INVITE BC2: unrestricted digital information with tones and announcements	

6.2.2.4 Sending of the CPG message

TP304001	SIP reference	e: RFC 3261 [6]		ISUP reference:	
			E	S 283 027 [1], clause 7.2.3.2.6	
TSS reference:	ISUP-SIP /Basic call/	Sending of the CPG mes	ssage		
SIP selection	PICS 3/1				
criteria:					
ISUP selection	PICS 4/9				
criteria:					
Test purpose:	180 received, a CPG	is sent when an ACM wa	is sent be	efore	
	Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 180 Ringing message: Sends the CPG message with the with the event indicator set to "Alerting".				
SIP Parameter				-	
values:					
ISUP Parameter					
values:					
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→	→	INVITE	
		Ti/w2 expired			
	ACM(no indication)	←			
	CPG(Alerting)	←	←	180 Ringing	
	Ringing tone				
	ANM	←	←	200 OK INVITE	
			→	ACK	
		Conve	ersation		
		→	→	BYE	
	RLC	+	-	200 OK BYE	

TP304002	SIP reference: RFC 3261 [6]		ISUP reference:		
11 304002	On reference. Ki O 3201 [0]		S 283 027 [1], clause 7.2.3.2.6		
TSS reference:	ISUP-SIP /Basic call/ Sending of the				
SIP selection	PICS 3/1	<u> </u>			
criteria:					
ISUP selection	NOT PICS 4/15				
criteria:					
Test purpose:	ACM was sent after $T_{I/W1}$ expiry, a 1	83 is not interwork	red		
	,,				
	Ensure that the SUT, having sent a A	CM message with	called party status "no indication" after		
	T _{I/W1} expiry, on receipt of a 183 Sess	sion progress mes	sage:		
	 ISUP message is sent backward 	•			
SIP Parameter					
values:					
ISUP Parameter					
values:					
Comments:		SUT	SIP		
	IAM →				
	T_I/W	1 expiry			
	ACM ←	→	INVITE		
		←	183 Session Progress		
	Ringing tone				
	ANM ←	←	200 OK INVITE		
		→	ACK		
		Conversation			
	→	→	BYE		
	RLC ←	+	200 OK BYE		

TP304003	SIP reference: RI	FC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.1.4		
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message					
SIP selection	PICS 3/1 AND PICS 4/18		ssaye			
criteria:	1 100 3/1 AND 1 100 4/10	AND 1 100 4/19				
ISUP selection						
criteria:						
Test purpose:	ACM was sent after T _{I/W1}	expiry, a CPG is se	nt when a	183 is received contains a PSTN XML		
	ProgressIndicator #7					
				er the expiry of ToiW1, on receipt of ressIndicator "Terminating access		
	 sends the CPG mess 	age with the event i	ndicator	set to "progress".		
SIP Parameter	183 Session Progress;					
values:	PSTN XML ProgressIndi		access IS	DN"(#7)		
ISUP Parameter	CPG; event indicator: pr	ogress				
values:	BCI					
	interworking indicator: r ISUP indicator: ISUP use		unterea (J)		
	ISDN access indicator: "te		"ואח			
Comments:	ISUP/BICC	SUT	DIN	SIP		
Comments.	IAM =			OII		
	17 (17)	T _{I/W1} expiry				
	ACM •	.,	→	INVITE		
			7			
	(3)		-	183 Session progress		
	J. J. 101 111 19/		_	180 Ringing		
	_	ing tone	_	200 OK INVITE		
	ANM €		←	200 OK INVITE		
		Conv	-	ACK		
	-		ersation	BYE		
	RLC •		→	200 OK BYE		
	RLC	-		200 ON DIE		

TP304004	SIP reference:	RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.1.4	
TSS reference:	ISUP-SIP /Basic call/ S	Sending of the CPG mes		- 100 01. [.], oludo : 110	
SIP selection	PICS 3/1 AND PICS 4/		ougo		
criteria:		. •			
ISUP selection					
criteria:					
Test purpose:	ACM was sent after, a	183 contains a PSTN X	ML Progr	ressIndicator #7 was received	
				er the reception of the 183 Session	
				"Terminating access ISDN"(#7), on	
			NXML bo	dy containing the progress	
	descriptions "Terminat	ing access ISDN"(#7)			
	 sends the CPG message with the event indicator set to "Alerting". 				
SIP Parameter					
		ProgressIndicator "To			
values:		sIndicator "Terminating	access is	SDN (#7)	
values:	CPG; event indicator:	Alerting			
values.	interworking indicator: no interworking encountered (0)				
	ISUP indicator: ISUP used all the way				
	ISDN access indicator: "terminating access ISDN"				
Comments:	ISUP/BICC	SUT	<u> </u>	SIP	
	IAM	→	→	INVITE	
	ACM	←	+	183 Session progress	
	CPG	-	←	180 Ringing	
		tinging tone	-		
	ANM	←	(200 OK INVITE	
			→	ACK	
		Conv	ersation	-	
		→	→	BYE	
	RLC	←	-	200 OK BYE	

TP304005	SIP reference: RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.1.4	
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG me		5 200 027 [1], 01443C 7.2.0.1.4	
SIP selection	PICS 3/1 AND PICS 4/18	oougo		
criteria:				
ISUP selection				
criteria:				
Test purpose:	ACM was sent after T _{I/W1} expiry, after receipt of a 183 and 180 contains a PSTN XML			
	ProgressIndicator #7 a CPG(Progress) and a	CPG(Alera	ting) are sent	
	Ensure that the SUT, having sent the ACM m			
	183 Session progress message followed by a containing the progress descriptions "Termina			
	Containing the progress descriptions Termina	alling acces	33 13DN (#1),	
	 sends two CPG messages respectively v 	vith the eve	ent indicator set to "Progress" and	
	"Alerting".			
SIP Parameter	183 Session progress			
values:	180 Ringing			
ISUP Parameter	CPG 1; event indicator: Progress			
values:	CPG 2; event indicator: Alerting			
	BCI:	untorod		
	interworking indicator: no interworking enco	Juntered		
	ISDN access indicator: "terminating access	ISDN"		
Comments:	ISUP/BICC SUT		SIP	
	IAM →			
	T _{I/W1} expiry			
	ACM(no indication) ←	→	INVITE	
	CPG(Progress)	←	183 Session progress	
	CPG(Alerting)	←	180 Ringing	
	Ringing tone			
	ANM ←	(200 OK INVITE	
		→	ACK	
		ersation		
	→	→	BYE	
	RLC ←		200 OK BYE	

TP304006	SIP reference:	RFC 3261 [6]		ISUP reference: S 283 027 [1], clause 7.2.3.1.4
TSS reference:	ISLIP-SIP /Basic call/ 9	Sending of the CPG med		.5 203 027 [1], clause 7.2.5.1.4
SIP selection	ISUP-SIP /Basic call/ Sending of the CPG message PICS 3/1 AND PICS 4/18			
criteria:	1 100 0/ 1 / 1110 1/			
ISUP selection criteria:				
Test purpose:	ACM was sent after T_{l}	_{/W1} expiry, a 183 cover	ing a PS7	N XML ProgressIndicator #7 and #x
	received, a CPG is ser	nt contains an ATP with	PI #x	
				ter the expiry of Ti/W1, on receipt of
				containing the progress descriptions gressIndicator #7 is not sent in the
	ATP.	DDN (#1) and FI_VALO	L lile Fio	gressindicator #7 is not sent in the
	71111			
	sends the CPG me	essage with the event i	ndicator	set to "progress" and the ATP
		or set to PI_VALUE .		
SIP Parameter	183 Session Progress			
values:		dicator " <i>Terminating ac</i>	cess ISD	N"(#7)
ISUP Parameter	PI_VALUE CPG; event indicator: progress			
values:	ATP progress indicate			
Comments:	ISUP/BICC	SUT		SIP
	IAM	→		
		T _{I/W1} expiry		
	ACM(no indication)	←	→	
	CPG(Progress)	←	←	183 Session progress
	CPG(Alerting)	←	←	180 Ringing
		linging tone	,	000 01/ 101/175
	ANM	←	←	200 OK INVITE ACK
		Conv	ersation	AUN
		→	ersation	BYE
	RLC	+	<i>•</i>	200 OK BYE

Values and additional selection criteria for test purposes TP304006				
VA_01	VA_01 PI_VALUE = Call is not end-to-end ISDN (#1)			
VA_02	PI_VALUE = Destination address is non-ISDN (#2)			

TP304007	SIP referenc	e: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.1.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message			
SIP selection	PICS 3/1 AND PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	ACM was sent after	T _{I/W1} expiry, a 180 coveri	ing a PS1	N XML ProgressIndicator #7 and #x
	received, a CPG is se	ent contains an ATP with	PI #x	
	·			
				n receipt of an a 180 Ringing message
	with PSTN XML Prog	gressIndicator "Termina	ting acces	ss <i>ISDN"(#7)"</i> and " PI_VALUE ",
	sends a CPG me	essage with the event inc	di cator se	et to "Alerting" and the ATP including
OID Development		dicator set to " PI_VALUI	Ξ.	
SIP Parameter values:	180 Ringing;	almaliaatarı DL \/ALLIC		
ISUP Parameter	CPG; Event indicate	sIndicator: PI_VALUE		
values:	ATP progress indicate			
Comments:	ISUP/BICC	SUT		SIP
Comments.	IAM	→		SIF
	IAW	=		
		T _{OIW1} expiry		
	ACM(no indication)	←	→	INVITE
	CPG(Alerting)	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conve	ersation	
		→	→	BYE
	RLC	+	+	200 OK BYE

Values and additional selection criteria for test purposes TP304007					
VA_01	VA_01 PI_VALUE = Call is not end-to-end ISDN (#1)				
VA 02	PI_VALUE = Destination address is non-ISDN (#2)				

TP304008	SIP reference	e: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.1.4
TSS reference:	ISUP-SIP /Basic call	/ Sending of the CPG me		i i i i i i i i i i i i i i i i i i i
SIP selection criteria:	PICS 2/3 AND PICS 4/18			
ISUP selection criteria:	PICS 4/19			
Test purpose:	ACM was sent after	T _{I/M/1} expiry, a 180 cover	ing a PST	TN XML ProgressIndicator #7 and #x
	received, a CPG is s	ent contains an ATP with	PI #x	
	Ensure that the SUT having sent the ACM message, on receipt of a 183 Call Progress message containing the PSTN XML ProgressIndicator " <i>Terminating access ISDN</i> "(#7) and "Interworking has occurred and has resulted in a telecommunication service change (#5)", the PSTN XML BearerCapability set to BC_VALUE			
	the BC set to B0	C_VALUE and the progre	ess indica	set to "Progress", the ATP containing ator set to "Interworking has occurred hange (#5)" and the TMU set to
SIP Parameter	INVITE;			
values:		arer Capability: INVITE _		
	PSTN XML second	Bearer Capability: INVIT	E _BC2	
	400 Call Dragges P	CTN VML Drawnaadadii		amusuldas has seemus das das d
				erworking has occurred and has PSTN XML BearerCapability:
	BC_VALUE	illitatileation service chan	ge (#3). i	OTTO AIVIE Bearer Capability.
ISUP Parameter		ıdio3Kbit/s, G.711 A-law		
values:		Digital info T/A, G.711 A-I	aw	
	TMR: 64 kbit/s prefe	rred		
	TMR prime: Speech/audio3Kbit/s			
	CPG, event indicator: Progress			
	ATP BC: BC_VALUE			
	service change (#5)	ator: interworking has oc	curred ar	nd has resulted in a telecommunication
	TMU: TMU_VALUE			
Commontos	ISUP/BICC	SUT		SIP
Comments:	IAM	→		
		T _{OIW1} expiry		
	ACM	←	→	INVITE
	CPG	←	←	
	CPG	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
		→	→	BYE
	RLC		+	200 OK BYE

TP304009	SIP refere	nce: RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.	
TSS reference:	ISLIP-SIP /Basic c	all/ Sending of the CPG me		203 027 [1], clause 7.2.3.	
SIP selection	PICS 2/3 AND PIC		bougo		
criteria:					
ISUP selection	PICS 4/19				
criteria:					
Test purpose:	ACM was sent after T _{I/W1} expiry, Fallback occurs in the 180 Ringing				
	Ensure that the SUT in call having sent the ACM message, on receipt of an 180 Ringing message containing the PSTN XML BearerCapability set to BC_VALUE and the PSTN XML ProgressIndicator set to " <i>Terminating access ISDN"(#7) and</i> "Interworking has occurred and has resulted in a telecommunication service change (#5)"				
	the BC set to	BC_VALUE and the progre ted in a telecommunication	ss indica	set to "Alerting", the ATP containing ator set to "Interworking has occurred hange (#5)" and the TMU set to	
SIP Parameter	INVITE;				
values:		earer Capability: INVITE _			
		d Bearer Capability: INVIT	E _BC2		
	180 Ringing;		10		
		resulted in a telecommunic		SDN"(#7) and Interworking has	
			allon Serv	ice change (#5)	
ISUP Parameter	PSTN XML BeaereCapabilty: BC_VALUE IAM; USI: Speech/audio3Kbit/s, G.711 A-law				
values:	USI prime: Unrestr. Digital info T/A, G.711 A-law				
	TMR: 64 kbit/s preferred				
	TMR prime: Speech/audio3Kbit/s				
	CPG; event indicator: Alerting				
	ATP BC: BC_VALUE ATP progress indicator: Interworking has occurred and has resulted in a telecommunication				
	service change (#5		cuireu aii	d has resulted in a telecommunication	
	TMU: TMU_VALU				
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→			
		T _{OIW1} expiry			
	ACM	←	→	INVITE	
	CPG	←	←	180 Ringing	
		Ringing tone			
	ANM	←	←	200 OK INVITE	
			→	ACK	
			ersation		
		→	→	BYE	
	RLC			200 OK BYE	

TP304010	SIP refere	nce: RFC 3261 [6]		ISUP reference:	
TSS reference:	ISLID SID /Basis s	all/ Sending of the CPG me		S 283 027 [1], clause 7.2.3.1.4.1	
SIP selection	PICS 2/3 AND PIC		ssaye		
criteria:	1 100 2/0 / (100 1 10	70 4/10			
ISUP selection	PICS 4/19				
criteria:					
Test purpose:	ACM was sent after T _{I/W1} expiry, Fallback occurs in the 183 Session Progress				
	message containir "Terminating acce- telecommunication	Ensure that the SUT having sent the ACM message, on receipt of a 183 Session Progress message containing the BC SET to BC_VALUE and the PSTN XML ProgressIndicator set to " <i>Terminating access ISDN"(#7)</i> , "Interworking has occurred and has resulted in a telecommunication service change (#5)" and "In-band information or appropriate pattern is now available (#8)",			
	appropriate pa	attern is now available", the	ATP con	set to "In-band information or taining the BC set to BC_VALUE and courred and has resulted in a MU set to TMU_VALUE.	
SIP Parameter	INVITE;				
values:		earer Capability: INVITE _			
	183 Session Progr	d Bearer Capability: INVIT	E _BC2		
	PSTN XML	655 ,			
		r: Interworking has occurred	d and has	resulted in a telecommunication	
	service change (#5)				
	ProgressIndicator: In-band information or appropriate pattern is now available (#8) PSTN XML BeaererCapability: BC_VALUE				
ISUP Parameter	IAM; USI: Speech/	audio3Kbit/s, G.711 A-law			
values:	USI prime: Unrestr. Digital info T/A, G.711 A-law				
	TMR: 64 kbit/s preferred				
	TMR prime: Speech/audio3Kbit/s				
	CPG; event indicator: In-band information or appropriate pattern is now available				
	ATP BC: BC_VALUE				
			curred ar	nd has resulted in a telecommunication	
	service change (#5				
Comments:	TMU: TMU_VALU	SUT		SIP	
Comments.	IAM	→		Sir	
	17 (17)	T _{OIW1} expiry			
	ACM	· O(VV · · · · · · ·)	4	INVITE	
	CPG	÷	É	183 Session progress	
	CPG	←	←	180 Ringing	
		Ringing tone		5 5	
	ANM	←	←	200 OK INVITE	
			→	ACK	
			ersation		
	51.6)	→	BYE	
	RLC	<u> </u>	<u> </u>	200 OK BYE	

Values and additional selection criteria for test purposes TP304008 to TP304010				
VA_01 TMU_VALUE: speech BC_VALUE: speech				
	ISUP_VALUE: UDI/TA	·		
VA_02	TMU_VALUE: 3,1 kHz	BC_VALUE: 3,1 kHz		
	ISUP_VALUE: UDI/TA			

TP3040011	SIP reference: RFC 3261 [6]		ISUP reference:	
TSS reference:	ISLID SID /Basic call/ Sanding of the CDC ma		S 283 027 [1], clause 7.2.3.1.4.1	
SIP selection	ISUP-SIP /Basic call/ Sending of the CPG message PICS 4/18			
criteria:	1 100 4/10			
ISUP selection				
criteria:				
Test purpose:	ACM was sent after T _{I/W1} expiry, PSTN XML HLC received in a 183 mapping in the ATP			
	contained in the CPG			
SIP Parameter values:	Ensure that the SUT, having sent the ACM message, on receipt of a 183 Session Progress message with PSTN XML ProgressIndicator set to "Interworking has occurred and has resulted in a telecommunication service change (#5)", "Terminating access ISDN"(#7) and with a PSTN XML HighLayerCompatibility set to HLC_VALUE the ProgressIndicator #7 is not contained in the ATP • sends the CPG message with event indicator set to "Progress", the ATP including the HLC set to HLC_VALUE and the progress indicator set to "Interworking has occurred and has resulted in a telecommunication service change (#5)". 183 Session Progress; PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a			
	telecommunication service change (#5) and "Terminating access ISDN"(#7)			
ISUP Parameter	PSTN XML HighLayerCompatibility: HLC_VALUE2 (PIXIT) CPG, Event indicator: Progress			
values:	ATP HLC: HLC_VALUE2 (PIXIT)			
10.000	ATP progress indicator: Interworking has occurred and has resulted in a telecommunication			
	service change (#5)			
Comments:	ISUP/BICC SUT		SIP	
	IAM →			
	T _{OIW1} expiry	/		
	ACM ←	→	INVITE	
	CPG ←	←	183 Session progress	
	CPG ←	←	180 Ringing	
	Ringing tone			
	ANM ←	←	200 OK INVITE	
	_	→	ACK	
		ersation	5)/5	
	→	→	BYE	
	RLC ←	<u>+</u>	200 OK BYE	

TP3040012	SIP reference: RFC 3261 [6]	ISUP reference:		
700 /		ES 283 027 [1], clause 7.2.3.1.4		
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message			
SIP selection	PICS 4/18			
criteria:	_			
ISUP selection				
criteria:	1014	1# 0		
Test purpose:	ACM was sent after $T_{I/W1}$ expiry, PSTN XML	HLC received in a 180 mapping in the ATP		
	contained in the CPG			
		essage, on receipt of an 180 Ringing message		
	with PSTN XML ProgressIndicator set to "Inte			
		minating access ISDN"(#7)and with a PSTN XML		
	HighLayerCompatibility set to HLC_VALUE			
	a sanda the CDC massage with event indi	cator set to "Alert", the ATP including the HLC		
		dicator set to "Interworking has occurred and		
	has resulted in a telecommunication serv			
SIP Parameter	180 Ringing;	ice change (#3) .		
values:	PSTN XML ProgressIndicator: Interworking	has occurred and has resulted in a		
varaco.	telecommunication service change (#5), " <i>Terminating access ISDN</i> "(#7)			
	PSTN XML HighLayerCompatibility: HLC_\			
ISUP Parameter	CPG, Event indicator: Alerting			
values:	ATP HLC: HLC_VALUE2 (PIXIT)			
	ATP progress indicator: Interworking has or	ccurred and has resulted in a telecommunication		
	service change (#5)			
Comments:	ISUP/BICC SUT	SIP		
	IAM →			
	T _{OIW1} expiry	1		
	ACM ←	→ INVITE		
	CPG ←	← 180 Ringing		
	Ringing tone	. 55		
	ANM •	← 200 OK INVITE		
		→ ACK		
	Conv	versation		
	→	→ BYE		
	RLC +	€ 200 OK BYE		
	1.120	- 200 010 01		

TP3040013	SIP reference: RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.1.4		
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG me		.5 205 027 [1], Clause 7.2.5.1.4		
SIP selection		ssaye			
criteria:					
ISUP selection	PICS 4/9				
criteria:					
Test purpose:	ACM sent after INVITE was sent, 183 receive	d after 18	0 received. 183 contains a PSTN XML		
• •	ProgressIndicator #7 mapped into BCI in the				
	Ensure that the SUT, having sent automatical				
	Ringing message followed by a 183 Session I		message with PSTN XML		
	ProgressIndicator "Terminating access ISDI	V"(#7),			
	and two CDC managers recorded to the	41-41	nt in diapton and to UA lautinall and		
	 sends two CPG message respectively wi "Progress". 	in the eve	ant indicator set to Alerting and		
SIP Parameter	180 Ringing;				
values:	183 Session Progress ;				
ISUP Parameter	CPG 1; event indicator: Alerting				
values:	ISUP indicator: ISUP is used all the way				
	ISDN access indicator: ISDN				
	Interwoking indicator: Interworking not enountered				
	CPG 2; event indicator: Progress				
	ISUP indicator: ISUP is used all the way				
	ISDN access indicator: ISDN				
Comments:	Interwoking indicator: Interworking not enount ISUP/BICC SUT	erea	SIP		
Comments:	IAM →		SIF		
	ACM ←	→	INVITE		
	ICPG ←	-			
	CPG ←	-	183 Session progress		
	Ringing tone	•	100 Ocasion progress		
	ANM ←	←	200 OK INVITE		
	7 (1)	À	ACK		
	Conv	ersation	7.01		
	REL →	→	BYE		
	RLC +	É	200 OK BYE		
L	1				

TP3040014	SIP reference: RFC 3261 [6]	ES	ISUP reference: 3 283 027 [1], clause 7.2.3.1.4	
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message			
SIP selection criteria:				
ISUP selection criteria:	PICS 4/9			
Test purpose:	ACM sent after INVITE was sent, a 180 is received contains a PSTN XML ProgressIndicator #x mapped into an PI #x covered in an ATP in the sent CPG Ensure that the SUT, having sent automatically the ACM message, on receipt of an 180 Ringing message containing the PSTN XML ProgressIndicator set to PI_VALUE, sends a CPG message with the event indicator set to "Alerting" and the ATP including the progress indicator set to PI_VALUE.			
SIP Parameter values:	180 Ringing; progress indicator: PI_VALUE			
ISUP Parameter values:	CPG; Event indicator: Alerting ATP progress indicator: PI_VALUE			
Comments:	SUT		SIP	
	IAM →			
	ACM ←	→	INVITE	
	CPG ←	←	180 Ringing	
	Ringing tone			
	ANM ←	←	200 OK INVITE	
	Conv	→ ersation	ACK	
	REL →	→	BYE	
	RLC ←	←	200 OK BYE	

Values and additional selection criteria for test purposes TP3040014				
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)			
VA 02	PL VALUE = Destination address is non-ISDN (#2)			

TP3040015	SIP referen	ce: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.1.4	
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message				
SIP selection					
criteria:					
ISUP selection	PICS 4/9				
criteria:					
Test purpose:	ACM sent after INVITE was sent, 180 received contains a PSTN XML ProgressIndicator #8 mapped into OBCI in the sent CPG Ensure that the SUT, having received an IAM with the USI field indicating USI_VALUE and having sent automatically the ACM message, on receipt of an 180 Ringing message with PI				
	No.8 "In-band inforr	nation or appropriate patte	rn is now	available",	
	 sends a CPG message with the event indicator set to "Alerting" and OBCI in-band information set to "yes". 				
SIP Parameter	180 Ringing; PSTN	XML ProgressIndicator"In	-band info	rmation or appropriate pattern is now	
values:	available" (#8)				
ISUP Parameter	IAM: USI : USI_VAL				
values:	CPG; Event indica	t or : Alerting			
	OBCI in-band: yes				
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→			
	ACM	←	→	INVITE	
	CPG	←	←	180 Ringing	
		Ringing tone			
	ANM	~	←	200 OK INVITE	
			→	ACK	
		Conv	ersation		
		→	→	BYE	
	RLC	+	+	200 OK BYE	

TP3040016	SIP reference: RFC 32	61 [6]		ISUP reference: ES 283 027 [1], clause	
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message				
SIP selection	Tool on Abasic call Gertaing C	or the or o messa	gc		
criteria:					
ISUP selection	PICS 4/9				
criteria:					
SIP Parameter values: ISUP Parameter values:	ACM sent after INVITE was sent, 180 received contains a P-Early-Media header mapped into the OBCI "inband info available" Ensure that the SUT, having received an IAM with the USI field indicating USI_VALUE and having sent automatically the ACM message, on receipt of an 180 Ringing message with P-Early-Media header authorizing early media", • sends a CPG message with the event indicator set to "Alerting" and OBCI in-band information set to "yes". 180 Ringing; P-Early-Media header authorizing early media IAM: USI: USI_VALUE; CPG; Event indicator: Alerting				
Comments:	OBCI in-band: yes	SUT		SIP	
Johnnents.	IAM →	551		5 11	
	ACM ←		→	INVITE	
	CPG ← Ringing to	ne	←	180 Ringing	
	ANM ←		←	200 OK INVITE	
			→	ACK	
		Convers	ation		
	→		→	BYE	
	RLC ←		+	200 OK BYE	

Values and additional selection criteria for test purposes TP304016					
VA_01	VA_01 USI_VALUE = speech				
VA_02	USI_VALUE = 3,1 kHz				

6.2.2.5 Sending of the ANM message

TP305001	SIP reference: RFC 3261 [6]		ISUP reference:		
		E	S 283 027 [1], clause 7.2.3.2.7a		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	An ANM is sent after a 200 OK INVITE is red	eived			
	Ensure that the SUT having sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running): Send ANM as determined by BICC/ISUP procedures. Stop any existing awaiting answer indication (e.g. ringing tone).				
SIP Parameter	200 OK INVITE;	<u> </u>	, ,		
values:	·				
ISUP Parameter values:	ANM;				
Comments:	ISUP/BICC SUT		SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	Ringing tone				
	ANM ←	←	200 OK INVITE		
		→	ACK		
	Conve	ersation			
	REL →	→	BYE		
	RLC ←	←	200 OK BYE		

TP305002	SIP reference: RFC 3261 [6]		ISUP reference:		
			ES 283 027 [1], clause 7.2.3.1.5		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Ans	wer Message	e (ANM)/		
SIP selection	PICS 4/18				
criteria:					
ISUP selection criteria:					
Test purpose:	An ANM is sent after a 200 OK INVITE is received. PSTN XML ProgressIndicator #x mapped into the ATP in the ANM				
	Ensure that the SUT, having sent the ACM message, on receipt of a 200 OK message containing the PSTN XML ProgressIndicator set to PI_VALUE • sends the ANM message with the ATP including the PSTN XML ProgressIndicator				
	set to PI_VALUE.	J	-		
SIP Parameter	200 OK; PSTN XML ProgressIndicator	: PI_VALUE	(PIXIT)		
values:	ANIA ATT B	(DI) (IT)			
ISUP Parameter values:	ANM; ATP Progress Indicator: PI_VAL	UE (PIXII)			
Comments:	ISUP/BICC SU	Т	SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	Ringing tone				
	ANM ←	←	200 OK INVITE		
		→	ACK		
	Conversation				
	REL →	→	BYE		
	RLC ←	←	200 OK BYE		

TP305003	SIP reference: RFC 3261 [6]	50	ISUP reference:		
			283 027 [1], clause 7.2.3.1.5		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer N	essage (A	.NM)/		
SIP selection	PICS 4/18				
criteria:					
ISUP selection					
criteria:					
Test purpose:	An ANM is sent after a 200 OK INVITE is	received. I	PSTN XML LowLayerCompatibility		
	mapped into the ATP in the ANM				
	Ensure that the SUT, having sent the ACM me				
	containing the PSTN XML LowLayerCompat	bility set t	to LLC_VALUE		
	 sends the ANM message with the ATP in 	cluding the	ELLC set to LLC_VALUE.		
SIP Parameter	200 OK; PSTN XML LowLayerCompatibility	LLC_VAL	LUE (PIXIT)		
values:					
ISUP Parameter	ANM; ATP LLC: PI_VALUE (PIXIT)				
values:					
Comments:	ISUP/BICC SUT	S	IP		
	IAM →	→ IN	IVITE		
	ACM ←	← 18	80 Ringing		
	Ringing tone				
	ANM ←	← 20	00 OK INVITE		
		→ A	CK		
	Conve	sation			
	REL →	→ B'	YE		
	RLC ←	← 20	00 OK BYE		

TP305004	SIP reference: RFC 3261 [6]		ISUP reference:		
			ES 283 027 [1], clause 7.2.3.1.5		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer M	lessage	e (ANM)/		
SIP selection	PICS 4/18 AND PICS 2/3				
criteria:					
ISUP selection	PICS 4/19				
criteria:					
Test purpose:	An ANM is sent after a 200 OK INVITE is i	eceive	d. PSTN XML HighLayerCompatibility		
	mapped into the ATP in the ANM				
	Ensure that the SUT, having sent the ACM me				
	containing the PSTN XML HighLayerCompat	ibility	set to HLC_VALUE1		
	Lat ANNA State ATTS:		# 111.6 ** 111.0 \\All.1154		
OID D	sends the ANM message with the ATP including the HLC set to HLC_VALUE1.				
SIP Parameter	200 OK; PSTN XML HighLayerCompatibility	: HLC_	VALUE2 (PIXIT)		
values:	LANCE ATT THE CALL HE CONCERN TO THE				
ISUP Parameter	IAM; ATP HLC1: HLC_VALUE1 (PIXIT)				
values:	ATP HLC2: HLC_VALUE2 (PIXIT)				
	ANM; ATP HLC: HLC_VALUE2 (PIXIT)				
Comments:	ISUP/BICC SUT		SIP		
Comments.	IAM →	→	INVITE		
		-			
	7.00 1.00 1.00 1.00 1.00 1.00 1.00 1.00				
	Ringing tone				
	ANM ←	(200 OK INVITE		
		→	ACK		
	Convei		5)/5		
	REL -	→	BYE		
	RLC ←		200 OK BYE		

TP305005	SIP referenc	e: RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.1.5		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/					
SIP selection criteria:	PICS 4/18 AND PICS 2/3					
ISUP selection	PICS 4/19					
criteria: Test purpose:	An ANIA is sent of	tor a 200 OK INVITE	io rooo	ived. PSTN XML BearerCompatibility		
rest purpose.		P and TMU in the ANM	s recei	ved. FSTN XIVIL BearerCompatibility		
		ge, on receipt of a 200 O		age indicating BC fallback and having sage with PSTN XML		
	sends the ANM TMU set to TMU		cluding	the BC set to BC_VALUE and the		
SIP Parameter	INVITE;					
values:		rer Capability: INVITE				
		Bearer Capability: INVIT	IF _BC	2		
	200 OK; PSTN XML BeaererCapability: BC_VALUE					
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law					
values:	USI prime: Unrestr. Digital info T/A, G.711 A-law					
	TMR: 64 kbit/s prefer					
	TMR prime: Speech/audio3Kbit/s					
		/A.L.I.E				
	ANM; ATP BC : BC_\ TMU : USI_VALUE	VALUE				
Comments:	ISUP/BICC	SUT		SIP		
Gommonto.	IAM	→	→	INVITE		
	ACM	-	-	180 Ringing		
		Ringing tone				
	ANM	←	←	200 OK INVITE		
			→	ACK		
		Conve	rsation			
	REL	→	→	BYE		
	RLC		+	200 OK BYE		

Values and additional selection criteria for test purposes TP TP305005					
VA_01	VA_01 TMU_VALUE: speech ISUP_VALUE: UDI/TA BC_VALUE: speech				
VA_02	TMU_VALUE: 3,1 kHz ISUP_VALUE: UDI/TA	BC_VALUE: 3,1 kHz			

TP305006	SIP reference: RFC 3261	[6]	i	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/					
SIP selection criteria:	PICS 4/18 AND PICS 2/3					
ISUP selection criteria:	PICS 4/19					
Test purpose:	An ANM is sent after a 200 OK INVITE is received. No PSTN XML BearerCompatibility contained in the 200. Sending of TMU in the ANM Ensure that the SUT, having received the IAM message indicating BC fallback and having sent the ACM message, on receipt of a 200 OK Message without PSTN XML BeaererCapability,					
	 sends the ANM message with TMU set to TMU_VALUE. 	the ATP in	cluding	the BC set to USI_VALUE and the		
SIP Parameter	INVITE;					
values:	PSTN XML first Bearer Capability: INVITE _BC1 PSTN XML second Bearer Capability: INVITE _BC2 200 OK; no BC					
ISUP Parameter values:	IAM; USI: Speech/audio3Kbit/s, G.711 A-law USI prime: Unrestr. Digital info T/A, G.711 A-law TMR: 64 kbit/s preferred TMR prime: Speech/audio3Kbit/s ANM; ATP BC: USI_VALUE TMU: USI_VALUE					
Comments:	ISUP/BICC	SUT		SIP		
	IAM →		→	INVITE		
	ACM ← Ringing tone		←	180 Ringing		
	ANM ←		←	200 OK INVITE		
			→	ACK		
		Conver				
	REL →		→	BYE		
	RLC ←			200 OK BYE		

Values and additional selection criteria for test purposes TP305006			
VA_01 TMU_VALUE: speech			
ISUP_VALUE: UDI/TA			
VA_02	TMU_VALUE: 3,1 kHz audio		
	ISUP_VALUE: UDI/TA		

TP305007	SIP reference: RFC 3261 [6]	F	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/				
SIP selection	PICS 4/18 AND PICS 2/3				
criteria:					
ISUP selection	PICS 4/19				
criteria:					
Test purpose:	An ANM is sent after a 200 OK INVITE is rece ProgressIndicator #5, mapped into the ATP in				
	Ensure that the SUT, having sent the ACM me containing the PSTN XML HighLayerCompat XML ProgressIndicator set to "Interworking helecommunication service change (#5)"	ibility	set to HLC_VALUE1 and the PSTN		
	 sends the ANM message with the ATP in the Progress Indicator set to "Interworking telecommunication service change (#5)". 				
SIP Parameter	200 OK INVITE				
values:	PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT)				
	PSTN XML ProgressIndicator: Interworking I	nas occ	curred and has resulted in a		
IOUD D	telecommunication service change (#5)				
ISUP Parameter values:	IAM; ATP HLC1: HLC_VALUE1 (PIXIT) ATP HLC2: HLC VALUE2 (PIXIT)				
values:	TIP HEOZ. HEO_VALUEZ (FIAIT)				
	ANM; ATP HLC: HLC_VALUE1 (PIXIT)				
	ATP Progress Indicator: Interworking has occurred and has resulted in a				
	telecommunication service change (#5)				
Comments:	ISUP/BICC SUT		SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	Ringing tone				
	ANM ←	←	200 OK INVITE		
		→	ACK		
	Conver	sation			
	REL →	→	BYE		
	RLC ←	+	200 OK BYE		

TP305008	SIP refere	ence: RFC 3261 [6]		ISUP reference:	
				ES 283 027 [1], clause 7.2.3.1.5	
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/				
SIP selection criteria:	PICS 4/18 AND PICS 2/3				
ISUP selection	PICS 4/19				
criteria:					
Test purpose:		after a 200 OK INVITE #5, mapped into the ATP		red. PSTN XML BearerCapability and J in the ANM	
	sent the ACM mes	ssage, on receipt of a 200 information element set	OK Mess to BC_VA	ALUE and the PSTN XML	
		or set to "Interworking has n service change (#5)"	occurred	d and has resulted in a	
	TMU set to U		ess Indic	the BC set to BC_VALUE and the cator set to "Interworking has occurred change (#5)".	
SIP Parameter	INVITE:				
values:		Bearer Capability: INVITE	BC1		
	PSTN XML second Bearer Capability: INVITE _BC2				
	200 OK:				
		erCapability: BC_VALUE			
			g has occ	curred and has resulted in a	
	telecommunication service change (#5)				
ISUP Parameter	IAM; USI : Speech/audio3Kbit/s, G.711 A-law				
values:	USI prime: Unrestr. Digital info T/A, G.711 A-law				
	TMR: 64 kbit/s preferred				
	TMR prime: Speech/audio3Kbit/s				
	ANM; ATP BC: BC_VALUE				
	ATP Progress Indicator: Interworking has occurred and has resulted in a				
	telecommunication service change (#5)				
	TMU: TMU_VALUE				
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→	→	INVITE	
	ACM	←	←	180 Ringing	
		Ringing tone		3 3	
	ANM	←	←	200 OK INVITE	
			→	ACK	
		Conv	ersation		
	REL	→	→	BYE	
	RLC	←	←	200 OK BYE	

Values and additional selection criteria for test purposes TP305008				
VA_01 TMU_VALUE: speech BC_VALUE: speech				
	ISUP_VALUE: UDI/TA			
VA_02	TMU_VALUE: 3,1 kHz	BC_VALUE: 3,1 kHz		
	ISUP_VALUE: UDI/TA			

6.2.2.6 Sending of the CON message

TP306001	SIP reference: RFC 3261 [6]	_	ISUP reference:	
			S 283 027 [1], clause 7.2.3.1.5	
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/			
SIP selection				
criteria:				
ISUP selection	NOT PICS 4/9			
criteria:				
Test purpose:	CON is sent after 200 was received			
	- 11 - 11 - OUT 1 - 1 - 1 - 1 - 1 - 1 - 1 - 1 - 1 - 1			
	Ensure that the SUT, having not sent the ACN		age, on receipt of a 200 OK INVITE	
	for this call, it shall stop timer TOIW2 (if runnir	g).		
	Send CON as determined by BICC/ISUP	nroced	ures	
	 Stop any existing awaiting answer indicat 			
	followed:	on (0.9	. Iniging tono, bor oncoded do	
	Interworking indicator: interworking	encou	ntered	
	ISUP indicator: ISUP not used all the way			
	ISDN access indicator: terminating access non-ISDN			
SIP Parameter	200 OK INVITE;			
values:				
ISUP Parameter	CON: Interworking indicator: interworking e		ered	
values:	ISUP indicator: ISUP not used all th			
	ISDN access indicator: terminating access non-ISDN			
Comments:	ISUP/BICC SUT		SIP	
	IAM →	→	INVITE	
	CON ←	←	200 OK INVITE	
		→	ACK	
	Conve	sation		
	REL →	→	BYE	
	RLC ←	←	200 OK BYE	

TP306002	SIP referen	ce: RFC 3261 [6]		ISUP reference:
				ES 283 027 [1], clause 7.2.3.1.5
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/			
SIP selection criteria:				
ISUP selection criteria:	NOT PICS 4/9 AND PICS 4/17			
Test purpose:	IAM received with T	MR 64 kBit/s. BCI in the C	CON inc	licates ISDN access
		Γ, having not sent the ACN stop timer TOIW2 (if running)		age, on receipt of a 200 OK INVITE
	 Send CON as of 	determined by BICC/ISUP	proced	ures.
	Stop any existing awaiting answer indication (e.g. ringing tone) BCI encoded as follows:			
	interworking indicator: no interworking encountered (0)			
	ISUP indicator: ISUP used all the way			
OID D	ISDN access indicator: "terminating access ISDN"			
SIP Parameter values:	200 OK INVITE			
ISUP Parameter	CON:			
values:		king indicator: no interwo		ncountered (0)
		licator: ISUP used all the		
_		cess indicator: "terminating	ng acce	
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	CON	←	←	200 OK INVITE
			→	ACK
		Conve	rsation	
	REL	→	→	BYE
	RLC	+	+	200 OK BYE

TP306003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7	.2.3.1.5
TSS reference:	ISUP-SIP/Basic call/ Sending of the Conne	t Message (CON)/	
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:			
Test purpose:	A CON is sent after a 200 OK INVITE is received. PSTN XML ProgressIndicator #x mapped into the ATP in the CON Ensure that on receipt of a 200 OK message containing the PSTN XML ProgressIndicator set to PI_VALUE sends the CON message with the ATP including the PSTN XML ProgressIndicator set to PI_VALUE.		
SIP Parameter values:	200 OK INVITE: PSTN XML ProgressIndi	ator: PI_VALUE (PIXIT)	
ISUP Parameter values:	ANM: ATP Progress Indicator: PI_VALUE	(PIXIT)	
Comments:	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
	CON ←	◆ 200 OK INVITE	
		→ ACK	
	Cor	ersation	
	REL →	→ BYE	
	RLC ←	★ 200 OK BYE	

TP306004	SIP reference: RFC 32	61 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.1.5
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/			
SIP selection	PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	A CON is sent after a 200 C mapped into the ATP in the CO		eceive	d. PSTN XML LowLayerCompatibility
	Ensure that on receipt of a 200 OK Message containing the PSTN XML			
	LowLayerCompatibility set to LLC_VALUE			
				
	sends the CON message with the			
SIP Parameter	200 OK INVITE: PSTN XML Lo	wLayerCompa	tibilit	y: LLC_VALUE (PIXIT)
values:		\		
ISUP Parameter	CON: ATP LLC : PI_VALUE (PI	XII)		
values:	IOUR/DIGO	OUT		OID
Comments:	ISUP/BICC	SUT		SIP
	IAM →		→	
	CON ←		←	200 OK INVITE
			→	ACK
		Convers	sation	
	REL →		→	BYE
	RLC ←		←	200 OK BYE

TP306005	SIP reference: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.1.5
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect I	Messag	e (CON)/
SIP selection criteria:	PICS 4/18 AND PICS 2/3		
ISUP selection criteria:	PICS 4/19		
Test purpose:	A CON is sent after a 200 OK INVITE is r mapped into the ATP in the CON Ensure that on receipt of a 200 OK Message of HighLayerCompatibility set to HLC_VALUE	containii	
	sends the CON message with the ATP includi	ng the I	HLC set to HLC_VALUE1.
SIP Parameter values:	200 OK INVITE: PSTN XML HighLayerComp	atibilit	y: HLC_VALUE2 (PIXIT)
ISUP Parameter values:	IAM; ATP HLC1: HLC_VALUE1 (PIXIT) ATP HLC2: HLC_VALUE2 (PIXIT) ACON: ATP HLC: HLC_VALUE2 (PIXIT)		
Comments:	ISUP/BICC SUT		SIP
	IAM →	→	INVITE
	CON ←	←	200 OK INVITE
		→	ACK
	Conver	sation	
	REL →	→	BYE
	RLC ←	-	200 OK BYE

TP306006	SIP reference: RFC 3261 [6]	ISUP reference:		
17300000	Sir reference: RFC 3261 [6]	ES 283 027 [1], clause 7.2.3.1.5		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/			
SIP selection	PICS 4/18 AND PICS 2/3			
criteria:	FIGS 4/16 AND FIGS 2/3			
ISUP selection	PICS 4/19			
criteria:				
Test purpose:	A CON is sent after a 200 OK INVITE is received. PSTN XML BearerCompatibility mapped into the ATP and TMU in the CON			
	Ensure that the SUT, having received the IAM message indicating BC fallback on receipt of a 200 OK Message with PSTN XML BeaererCapability set to BC_VALUE, sends the CON message with the ATP including the BC set to BC_VALUE and the TMU			
	set to TMU_VALUE.			
SIP Parameter values:	INVITE; PSTN XML first Bearer Capability: INVITE _BC1 PSTN XML second Bearer Capability: INVITE _BC2 200 OK INVITE PSTN XML BeaererCapability: BC_VALUE			
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law			
values:	USI prime: Unrestr. Digital info T/A, G.711 A-law			
	TMR: 64 kbit/s preferred			
	TMR prime: Speech/audio3Kbit/s			
	CON: ATP BC: BC_VALUE			
	TMU: TMU_VALUE			
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	CON ←	← 200 OK INVITE		
		→ ACK		
	Conve	ersation		
	REL →	→ BYE		
	RLC ←	★ 200 OK BYE		

Values and additional selection criteria for test purposes TP TP306006			
VA_01	TMU_VALUE: speech	BC_VALUE: speech	
	ISUP_VALUE: UDI/TA		
VA_02	TMU_VALUE: 3,1 kHz	BC_VALUE: 3,1 kHz	
	ISUP VALUE: UDI/TA		

TP306007	SIP reference: RI	FC 3261 [6]		ISUP reference:
T00 (10115 015/5 : 11/ 0	F (4 0 4		ES 283 027 [1], clause 7.2.3.1.5
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/			
SIP selection	PICS 4/18 AND PICS 2/3			
criteria:				
ISUP selection criteria:	PICS 4/19			
	A CON is sent often a 3	OO OK INVITE is n	2001100	I. No PSTN XML BearerCompatibility
Test purpose:	contained in the 200. Sen			I. NO PSTN XIVIL BearerCompatibility
				age indicating BC fallback on receipt of
	a 200 OK Message without	ut PSTN XML Beaer	erCap	ability,
	sends the CON message	with the ATD includi	na the	BC set to USI_VALUE and the TMU
	set to TMU VALUE.	with the ATF includi	ilg tile	be set to osi_value and the find
SIP Parameter	INVITE:			
values:	PSTN XML first Bearer (Capability: INVITE	BC1	
	PSTN XML second Bearer Capability: INVITE _BC2			
	200 OK INVITE: no BC			
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law			
values:	USI prime: Unrestr. Digital info T/A, G.711 A-law			
	TMR: 64 kbit/s preferred	214111		
	TMR prime: Speech/audi	o3Kbit/s		
	CON: ATP BC: LIST VALUE	IE		
	CON: ATP BC: USI_VALUE TMU: TMU_VALUE			
Comments:	ISUP/BICC SUT SIP			
	IAM	→	→	INVITE
	CON	←	+	200 OK INVITE
		_	→	ACK
		Convei	sation	-
	REL	→	→	BYE
	RLC	←	←	200 OK BYE

Values and additional selection criteria for test purposes TP306007				
VA_01	TMU_VALUE: speech			
	ISUP_VALUE: UDI/TA			
VA_02	TMU_VALUE: 3,1 kHz audio			
ISUP_VALUE: UDI/TA				

TP306008	SIP reference: RFC 3261 [6]	ISUP reference:					
TSS reference:	ES 283 027 [1], clause 7.2.3.1.5						
SIP selection	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/						
criteria:	PICS 4/18 AND PICS 2/3						
ISUP selection	PICS 4/19						
criteria:	PICS 4/19						
Test purpose:	A CON is sent after a 200 OK INVITE is received. PSTN XML HighLayerCompatibility and ProgressIndicator #5, mapped into the ATP in the CON						
	Ensure that the SUT on receipt of a 200 OK Message containing the PSTN XML HighLayerCompatibility set to HLC_VALUE1 and the PSTN XML ProgressIndicator set						
	to "Interworking has occurred and has resulted in a telecommunication service change (#5)"						
	sends the CON message with the ATP including the HLC set to HLC_VALUE1 and the Progress Indicator set to "Interworking has occurred and has resulted in a						
	telecommunication service change (#5)".						
SIP Parameter	200 OK INVITE						
values:	PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT)						
	PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a						
	telecommunication service change (#5)						
ISUP Parameter	IAM; ATP HLC1: HLC_VALUE1 (PIXIT)						
values:	ATP HLC2: HLC_VALUE2 (PIXIT)						
	CON: ATP HLC: HLC_VALUE1 (PIXIT)	1 11 16 12					
	ATP Progress Indicator: Interworking has occurred and has resulted in a telecommunication service change (#5)						
Comments:	ISUP/BICC SUT	SIP					
Comments.	IAM →	→ INVITE					
	CON	€ 200 OK INVITE					
	CON	→ ACK					
	Conversation						
	REL →						
	RLC ← 200 OK BYE						

TP306009	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5		
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/			
SIP selection criteria:	PICS 4/18 AND PICS 2/3			
ISUP selection criteria:	PICS 4/19			
Test purpose:	A CON is sent after a 200 OK INVITE is received. PSTN XML BearerCapability and ProgressIndicator #5, mapped into the ATP and TMU in the CON			
	Ensure that the SUT, having received the IAM message indicating BC fallback and having sent the ACM message, on receipt of a 200 OK Message with PSTN XML BearerCapability information element set to BC_VALUE and the PSTN XML ProgressIndicator set to "Interworking has occurred and has resulted in a telecommunication service change (#5)"			
	sends the CON message with the ATP including the BC set to BC_VALUE and the TMU set to TMU_VALUE and the Progress Indicator set to "Interworking has occurred and has resulted in a telecommunication service change (#5)".			
SIP Parameter	INVITE;			
values:	PSTN XML first Bearer Capability: INVITE _BC1 PSTN XML second Bearer Capability: INVITE _BC2 200 OK INVITE PSTN XML BearerCapability: BC_VALUE PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a telecommunication service change (#5)			
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law			
values:	USI prime: Unrestr. Digital info T/A, G.711 A-law			
	TMR: 64 kbit/s preferred			
	TMR prime: Speech/audio3Kbit/s			
	CON: ATP BC: BC_VALUE			
	ATP Progress Indicator: Interworking has occurred and has resulted in a			
	telecommunication service change (#5)			
-	TMU: TMU_VALUE	OLD.		
Comments:	ISUP/BICC SUT	SIP → INVITE		
	IAM →	→ INVITE ← 200 OK INVITE		
	-	→ ACK		
	Conve	rsation		
	REL →	→ BYE		
	RLC ←	← 200 OK BYE		

Values and additional selection criteria for test purposes TP306009			
VA_01	TMU_VALUE: speech ISUP_VALUE: UDI/TA	BC_VALUE: speech	
VA_02	TMU_VALUE: 3,1 kHz ISUP_VALUE: UDI/TA	BC_VALUE: 3,1 kHz	

6.2.2.7 Receipt of the Release message (REL)

TP307001	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.8	
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	REL received before INVITE was sent		
CID Parameter	Ensure that the SUT after receiving the IAM but before an INVITE has been sent. On receipt of a REL message: no action is required on the SIP side other than to terminate local procedures if any are in progress.		
SIP Parameter values:			
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)		
Comments:	ISUP/BICC SUT	SIP	
	IAM →		
	REL →		
	RLC		

TP307002	SIP reference: RFC 3261 [6]	ISUP reference:			
TSS reference:	ES 283 027 [1], clause 7.2.3.1.8				
	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	REL received, BYE is sent after ACK for 200 OK was sent before early dialogue				
	 Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message on receipt REL message before a 200 OK response (any) message has been received which establishes a confirmed dialogue: The SUT shall hold the REL message until a SIP 200 OK INVITE response has been received. The SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP. 				
SIP Parameter values:	cause value: CV_SIP (PIXIT)				
ISUP Parameter	REL: cause value: CV_ISUP (PIXIT)				
values:	location: LOC_ISUP (PIXIT)				
Comments:	ISUP/BICC SUT	SIP			
	IAM →	→ INVITE			
	REL →				
	RLC ←				
		← 200 OK INVITE			
		→ ACK			
		→ BYF			
		€ 200 OK BYE			
		ZUU UN DIE			

OID/D : II/D : ((1		ES 2	283 027 [1], clause 7.2.3.1.8	
ISUP-SIP/Basic call/ Receipt of the Release message (REL)/				
•			·	
	0// 0 / / /			
200 OK INVITE received before 200 OK CANCEL was received. A BYE is sent Ensure that the SUT after receiving the IAM with the complete called party number,				
sending an INVITE message. On receipt of a REL message before a 200 OK response				
message has been received:				
The SUT shall hold the REL message. A CANCEL is sent when any SIP response was been received.				
 On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP 				
BYE: cause value: CV_SIP (PIXIT)				
	XIT)			
_ , ,				
	SUT	_	SIP	
→		_	INVITE	
•		+	100 Trying	
=				
←			CANOFI	
		_	CANCEL 200 OK INVITE	
		=	200 OK INVITE 200 OK CANCEL	
		_	ACK	
		_	BYE	
		-	200 OK BYE	
	e that the SUT after receiving an INVITE message. On age has been received: the SUT shall hold the REL as been received. In subsequently receiving 2 are 200 OK INVITE and subsent. The cause Value Indicate Reason header field deficause value: CV_SIP (PIX	e that the SUT after receiving the IAM wing an INVITE message. On receipt of a Fage has been received: the SUT shall hold the REL message. A Cas been received. In subsequently receiving 200 OK INVITE and subsequently sender. The cause Value Indicator paramete are Reason header field defined as CV_SICAUSE value: CV_SIP (PIXIT) Cause value: CV_ISUP (PIXIT) Cause value: CV_ISUP (PIXIT) BICC SUT	e that the SUT after receiving the IAM with the coming an INVITE message. On receipt of a REL message has been received: The SUT shall hold the REL message. A CANCEL is as been received. The subsequently receiving 200 OK INVITE message are 200 OK INVITE and subsequently send a BYE report. The cause Value Indicator parameter defined are Reason header field defined as CV_SIP. The cause value: CV_SIP (PIXIT) The cause value: CV_ISUP (PIXIT)	

TP307004	SIP reference: RFC 3261 [6]	ES 2	ISUP reference: 83 027 [1], clause 7.2.3.1.8
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	REL received before early dialogue is establication. Ensure that the SUT after receiving the IAM sending an INVITE message. On receipt of a the message defined as SIP_MESSAGE has The SUT shall hold the REL message ur received. The SUT shall send a CANCEL request defined as CV_ISUP shall be mapped to	with the com REL messa been estab htil a SIP_MI The cause	ge before an early dialogue with lished: ESSAGE_VA response has been Value Indicator parameter
SIP Parameter values:	CANCEL: cause value: CV_SIP (PIXIT)		
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)		
Comments:	ISUP/BICC SUT		SIP
	IAM REL → RLC ←	→	INVITE
		+ + + +	SIP_MESSAGE_VA CANCEL 200 OK CANCEL 487 Request terminated ACK

TP307005	SIP reference: RFC 3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.2.3.1.8		
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	 REL received, BYE is sent after ACK for 200 OK was sent in early dialogue Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message after a 200 OK response message has been received: The SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP. 				
SIP Parameter values:	BYE: cause value: CV_SIP (PIXIT)				
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)				
Comments:	ISUP/BICC SUT		SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	ANM ← REL →	+	200 OK INVITE		
	RLC ←	→	ACK		
		→	BYE		
		+	200 OK BYE		

TP307006	SIP reference: RFC 3261 [6]	ES 2	ISUP reference: 83 027 [1], clause 7.2.3.1.8	
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/			
SIP selection criteria:		g- (<i>_</i> -	
ISUP selection criteria:				
Test purpose:	 REL received, BYE is sent after ACK for 200 OK was sent in early dialogue established by several messages Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: The SUT shall send a CANCEL request. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP. 			
SIP Parameter	CANCEL: cause value: CV_SIP (PIXIT)	accii ilcaaci	noid doilined do OV_OII .	
values:				
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)			
Comments:	ISUP/BICC SUT		SIP	
	IAM →	→	INVITE	
		←	SIP_MESSAGE_VA	
	REL →			
	RLC ←			
		→	CANCEL	
		←	200 OK CANCEL	
		←	487 Request terminated	
		→	ACK	

Table 11

Values for test purpose TP307004; TP307006		
VA	VA SIP MESSAGE_VA	
VA_1	180 Ringing	
VA_2	181 Call Is Being Forwarded	
VA_3	182 Queued	
VA_4	183 Session Progress	

Table 12

	Values for test purposes 307004 - 307006				
←SIP Message Reason header field CV_SIP		← REL Cause Indicators parameter CV_ISUP			
VA_1	Normal call clearing # 16	Normal call clearing # 16			
VA_2	Normal, unspecified # 31	Normal, unspecified # 31			
VA_3	Temporary failure # 41	Temporary failure # 41			
VA_4	Invalid message, unspecified # 95	Invalid message, unspecified # 95			
VA_5	Recovery on timer expiry # 102	Recovery on timer expiry # 102			
VA_6	Protocol error, unspecified # 111	Protocol error, unspecified # 111			

Table 13: Mapping of Cause Indicators parameter into SIP Reason header fields

Cause indications parameter field	Value of parameter field	component of SIP Reason header field	Component value
-	-	Protocol	"ITU-T Rec Q.850 [5]"
Cause Value	"XX" (see note 1)	Protocol-cause	"cause= XX" (see note 1)
	-	Reason-text	Should be filled with the definition text as stated in ITU-T Recommendation Q.850 [5] (see note 2)

NOTE 1: "XX" is the Cause Value as defined in ITU-T Rec Q.850 [5].

NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in table1/ITU-T Recommendation Q.850 [5] this is based on provisioning in the O-IWU.

6.2.2.8 Sending of a REL message / receipt of a backward BYE

TP308001	SIP reference: RFC 3261		ISUP reference: ES 283 027 [1], clause 7.2.3.1.7		
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	BYE received, REL cause #16 is	sent			
	of a BYE message where a Reaso Cause Value is not included:	on header field with	ut an INVITE message and on receipt ITU-T Recommendation Q.850 [5] Illue No. 16 ("normal clearing").		
SIP Parameter			•		
values:					
ISUP Parameter values:	REL; Cause value "Normal call cle	earing"			
Comments:	ISUP/BICC	SUT	SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	Ringing tone				
	ANM ←	←	200 OK INVITE		
		→	ACK		
		Conversation			
	REL ←	←	BYE		
	RLC →	→	200 OK BYE		

TP308002	SIP reference: RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.1.7		
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release				
SIP selection	PICS 4/11		g- (<u>-</u>		
criteria:					
ISUP selection					
criteria:					
Test purpose:	BYE Reason header #x received, REL cause	#x is se	ent		
	Ensure that the SUT after receiving the IAM sends out a INVITE message and on receipt of a BYE message where a Reason header field with ITU-T Recommendation Q.850 [5] Cause Value is included: • sends a REL message. The Cause Value is in the Reason header filed mapped to the ISUP Cause Value field in the ISUP REL.				
SIP Parameter	BYE cause value: CV_SIP (PIXIT)	-			
values:					
ISUP Parameter	REL; cause value: CV_ISUP (PIXIT)				
values:					
Comments:	ISUP/BICC SUT		SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	_ ~ ~	Ringing tone			
	ANM ←	+	200 OK INVITE		
		→	ACK		
		rsation			
	REL ←	←	BYE		
	RLC →	→	200 OK BYE		

Table 14: Mapping of SIP Reason header fields into Cause Indicators parameter

component of SIP Reason header field	Component value	BICC/ISUP Parameter / field	value
Protocol	"ITU-T Rec Q.850 [5]"	Cause Indication parameter	-
protocol-cause	"cause = XX" (see note)	Cause Value	"XX" (see note)
-	-	Location	"network beyond interworking point"
NOTE: "XX" is the Ca	use Value as defined in ITI	J-T Recommendation Q.850 [5].	

TP308003	SIP reference:	RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.1.7	
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/				
SIP selection criteria:		<u> </u>		S- ()	
ISUP selection criteria:					
Test purpose:	Final response without	Reason header receive	ed, ma _l	oping in REL	
SIP Parameter values:	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included defined as SIP_Failure_VA: • sends a REL message with the Cause value set to CV_ISUP.				
ISUP Parameter	REL; cause value: CV	ISUP			
values:					
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→	→	INVITE	
	REL	(←	SIP_Failure_VA	
	RLC	→	→	ACK	

Table 15

Values for test purpose TP308003 VA ←REL (Cause Value) ←4XX/5XX/6XX SIP message				
*^	CV_ ISUP	SIP_Failure_VA		
VA_01	127 Interworking	400 Bad Request		
VA_02	127 Interworking	401 Unauthorised		
VA_03	127 Interworking	402 Payment Required		
VA_04	127 Interworking	403 Forbidden		
VA_05	127 Interworking	405 Method Not Allowed		
VA_06	127 Interworking	406 Not Acceptable		
VA_07	127 Interworking	407 Proxy authentication required		
VA_08	127 Interworking	408 Request Timeout		
VA_09	22 Number changed (without diagnostic)	410 Gone		
VA_10	127 Interworking	413 Request Entity too long		
VA_11	127 Interworking	414 Request-uri too long		
VA_12	127 Interworking	415 Unsupported Media type		
VA_13	127 Interworking	416 Unsupported URI scheme		
VA_14	127 Interworking	420 Bad Extension		
VA_15	127 Interworking	421 Extension required		
VA_16	127 Interworking	423 Interval Too Brief		
VA_17	20 Subscriber absent	480 Temporarily Unavailable		
VA_18	127 Interworking	481 Call/Transaction does not exist		
VA_19	127 Interworking	482 Loop Detected		
VA_20	127 Interworking	483 Too many hops		
VA_21	127 Interworking	485 Ambiguous		
VA_22	17 User busy	486 Busy Here		
VA_23	127 Interworking	488 Not acceptable here		
VA_24	127 Interworking	493 Undecipherable		
VA_25	127 Interworking	500 Server Internal error		
VA_26	127 Interworking	501 Not implemented		
VA_27	127 Interworking	502 Bad Gateway		
VA_28	127 Interworking	503 Service Unavailable		
VA_29	127 Interworking	504 Server timeout		
VA_30	127 Interworking	505 Version not supported		
VA_31	127 Interworking	513 Message too large		
VA_32	127 Interworking	580 Precondition failure		
VA_33	17 User busy	600 Busy Everywhere		
VA_34	21 Call rejected	603 Decline		
VA_35	1 Unallocated number	604 Does not exist anywhere		
VA_36	127 Interworking	606 Not acceptable		

TP308004	SIP reference:	RFC 3261 [6]		ISUP reference:
			E	ES 283 027 [1], clause 7.2.3.1.7
TSS reference:	ISUP-SIP /Basic call/ S	ending of the Release	messa	ge (REL)/
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Final response without	Reason header receive	ed, maj	oping in REL
				ut an INVITE message. On receipt of
	a Failure message (4xx	x, 5xx, 6xx) where a Re	ason h	eader field with
	ITU-T Recommendation	n Q.850 [5] Cause Valu	ie CV_	ISUP is included defined as
	SIP_Failure_VA:			
	 sends a REL message with the Cause value set to CV_ISUP. 			
SIP Parameter				
values:				
ISUP Parameter	REL; cause value: CV	_ISUP		
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	REL(#xx)	←	←	SIP_Failure_VA
	RLC	→	→	ACK

Table 16

	Values for test purpose TP308004, TP308005			
VA	←REL (Cause Value PIXIT)	←4XX/5XX/6XX SIP message SIP_Failure_VA		
VA_01	CV_ ISUP	400 Bad Request		
VA_02	CV_ ISUP	401 Unauthorised		
VA_03	CV_ISUP	402 Payment Required		
VA_04	CV_ ISUP	403 Forbidden		
VA_05	CV_ ISUP	405 Method Not Allowed		
VA_06	CV_ ISUP	406 Not Acceptable		
VA_07	CV_ ISUP	407 Proxy authentication required		
VA_08	CV_ ISUP	408 Request Timeout		
VA_09	CV_ ISUP	410 Gone		
VA_10	CV_ ISUP	413 Request Entity too long		
VA_11	CV_ ISUP	414 Request-uri too long		
VA_12	CV_ ISUP	415 Unsupported Media type		
VA_13	CV_ ISUP	416 Unsupported URI scheme		
VA_14	CV_ ISUP	420 Bad Extension		
VA_15	CV_ ISUP	421 Extension required		
VA_16	CV_ ISUP	423 Interval Too Brief		
VA_17	CV_ ISUP	480 Temporarily Unavailable		
VA_18	CV_ ISUP	481 Call/Transaction does not exist		
VA_19	CV_ ISUP	482 Loop Detected		
VA_20	CV_ ISUP	483 Too many hops		
VA_21	CV_ ISUP	485 Ambiguous		
VA_22	CV_ ISUP	486 Busy Here		
VA_23	CV_ ISUP	488 Not acceptable here		
VA_24	CV_ ISUP	493 Undecipherable		
VA_25	CV_ ISUP	500 Server Internal error		
VA_26	CV_ ISUP	501 Not implemented		
VA_27	CV_ ISUP	502 Bad Gateway		
VA_28	CV_ ISUP	503 Service Unavailable		
VA_29	CV_ ISUP	504 Server timeout		
VA_30	CV_ ISUP	505 Version not supported		
VA_31	CV_ ISUP	513 Message too large		
VA_32	CV_ ISUP	580 Precondition failure		
VA_33	CV_ ISUP	600 Busy Everywhere		
VA_34	CV_ ISUP	603 Decline		
VA_35	CV_ ISUP	604 Does not exist anywhere		
VA_36	CV_ ISUP	606 Not acceptable		

TP308005	SIP reference:	RFC 3261 [6]	_	ISUP reference: S 283 027 [1], clause 7.2.3.1.7
	7.7			
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/			
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Final response contains	s a Reason header in e	early dia	logue received
SIP Parameter	Ensure that the SUT after receiving the IAM sends out an INVITE message, a SIP message defined as SIP_MESSAGE_VA has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA where a Reason header field with ITU-T Recommendation Q.850 [5] Cause Value is included: • sends a REL message. The Cause Value in the header field set to CV_SIP is mapped to the ISUP Cause Value field in the ISUP REL message with the Cause value set to CV_ISUP.			
values:	CV_SIP (PIXIT)			
ISUP Parameter	CV_ ISUP (PIXIT)			
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
			←	SIP MESSAGE_VA
	REL	←	←	SIP Failure VA
	RLC	÷	÷	ACK
				7.011

Table 17

Values for test purpose TP308005			
VA	SIP MESSAGE_VA		
VA_1	180 Ringing		
VA_2	181 Call Is Being Forwarded		
VA_3	182 Queued		
VA_4	183 Session Progress		

TP308006	SIP reference: R	FC 3261 [6]	i	ISUP reference: ES 283 027 [1], clause 7.2.3.1.7
TSS reference:	ISUP-SIP /Basic call/ Se	nding of the Release	messa	ge (REL)/
SIP selection criteria:		· ·		
ISUP selection criteria:				
Test purpose:	Final response without R	Reason header receiv	ed, ma _l	oping in REL
	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a response message (3xx) defined as SIP_Response_VA, the SUT: • sends a REL message with the Cause value 127 Interworking.			
SIP Parameter values:				
ISUP Parameter values:	REL; cause value: 127			
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	REL(#127)	←	←	SIP_Response_VA
	RLC	<u>→</u>	<u>→</u>	ACK

Table 18

	Values for test purposes TP308006			
VA	←REL (Cause Value) CV_ISUP	←3XX SIP message SIP_Response_VA		
VA_01	127 Interworking	300 Multiple Choices		
VA_02	127 Interworking	301 Moved Permanently		
VA_03	127 Interworking	302 Move Temporarily		
VA_04	127 Interworking	305 Use Proxy		
VA_05	127 Interworking	380 Alternative Service		

6.2.2.9 Autonomous release at O-MGCF

TP309001	SIP reference: RFC 3	3261 [6]		ISUP reference:		
			ES 28	3 027 [1], clause 7.2.3.2.12.1		
TSS reference:	ISUP-SIP/Basic call/Autonom	ous release/				
SIP selection	PICS 3/2					
criteria:						
ISUP selection						
criteria:	0 1 17/0: 1					
Test purpose:	Overlap supported, REL is sent when 404/484 received and Ti/w3 is expired					
		Ensure that the SUT a On receipt of a 484 Address Incomplete or 404 Not Found				
				pending INVITE transactions for		
	SUT:	ired to propagate	e overlap s	ignalling into the SIP network, the		
	 Shall not send a REL me 	ssage immediate	ely and sha	all instead start timer TOIW3. The		
	REL message shall only The REL message conta					
SIP Parameter						
values:						
ISUP Parameter						
values:						
Comments:	ISUP/BICC	SUT	_	SIP		
	IAM →		→	INVITE		
	CASI	ĒΑ				
			(484 Address Incomplete		
		O T	→	ACK		
		Start timer T _{I/\}	N3			
		Timeout T _{I/W}	/3			
	REL #28 ←					
	RLC →					
	CASI	ЕВ				
			(404 Not Found		
		Start timer T _{I/\}	→ N3	ACK		
		Timeout T _{I/W}	/3			
	REL #28 ← RLC →					

TP309002	SIP reference: RFC	3261 [6]	ES 28	ISUP reference: 33 027 [1], clause 7.2.3.2.12.1
TSS reference:	ISUP-SIP/Basic call/Autonor	mous release/		[1],
SIP selection criteria:	NOT PICS 3/2	mods release/		
ISUP selection criteria:				
Test purpose:	Overlap not supported, REL	is sent when 404,	/484 receiv	/ed
	Ensure that the SUT on receipt of a 484 Address Incomplete response for the current INVITE (i.e. there are no other pending INVITE transactions for this call), if the O-MGCF is not configured to propagate overlap signalling into the SIP network then the timer shall not be started and the: • REL shall be sent immediately to the BICC/ISUP network.			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM →		→	INVITE
	REL #28 ←		←	484 Address Incomplete
	RLC →		→	ACK

TP309003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.18		
TSS reference:	ISUP-SIP/Basic call/Autonomous release/	1		
SIP selection	PICS 4/5 AND PICS 4/11			
criteria:				
ISUP selection criteria:	PICS 4/2			
Test purpose:	Preconditions supported, call setup released w	hen COT(failed) received		
	Ensure that the SUT a on receipt of a COT "failed" and preconditions used, the SUT: sends a CANCEL to the SIP network.			
SIP Parameter values:				
ISUP Parameter values:	IAM: Nature of connection indicators "continu	ity check required on this circuit"		
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	COT(failed) →	 → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK 		

TP309004	SIP reference: RFC 3261 [6]	ISUP reference: Es 283 027 [1], clause 7.7.3		
TSS reference:	ISUP-SIP/Basic call/Autonomous release/			
SIP selection criteria:	PICS 4/5 AND PICS 4/11			
ISUP selection criteria:	PICS 4/2			
Test purpose:	Preconditions supported, call setup released v	when T8 expired		
	Ensure that the SUT when the ISUP/BICC timer T8 is expired and preconditions used, the SUT:			
	 sends a CANCEL or BYE to the SIP netw 	ork.		
SIP Parameter values:				
ISUP Parameter values:	IAM: Nature of connection indicators "contin	uity check required on this circuit"		
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	T8 expirees			
		→ CANCEL		
		← 200 OK CANCEL		
		 487 Request terminated 		
	<u> </u>	→ ACK		

TP309005	SIP reference:	RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.16
TSS reference:	ISUP-SIP/Basic call/Autonomous release/			
SIP selection criteria:	PICS 4/7 AND PICS 4/1	15		
ISUP selection criteria:	PICS 4/2			
Test purpose:	Preconditions supported	d, 580 mapped in REL	#47	
	Ensure that the SUT when the resource reservation is unsuccessful and preconditions used, the SUT responds to an INVITE: • send a REL with cause value # 47			
SIP Parameter values:				
ISUP Parameter values:	IAM: Nature of connection indicators "continuity check required on this circuit"			
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	REL	←	←	580 Precondition Failure
	RLC	<u>→</u>	→	ACK

TP309006	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.17.2		
TSS reference:	ISUP-SIP/Basic call/Autonomous release/			
SIP selection criteria:	PICS 4/7 AND PICS 4/15			
ISUP selection criteria:	PICS 4/2			
Test purpose:	Preconditions supported, 580 mapped in REL	_ #47		
SIP Parameter	Ensure that the SUT when the resource reservation is unsuccessful and preconditions used, the SUT responds to an UPDATE: send a REL with cause value # 47			
values:				
ISUP Parameter values:	IAM: Nature of connection indicators "contin	nuity check required on this circuit"		
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
		 183 Session Progress 		
		→ PRACK		
		◆ 200 OK PRACK		
	REL ←	 580 Precondition Failure 		
	RLC →	→ ACK		

6.2.2.10 Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented

6.2.2.10.1 Receipt of Reset Circuit message (RSC)

TP310001	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15				
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit	message (RSC)				
SIP selection criteria:		•				
ISUP selection criteria:						
Test purpose:	RSC received while an INVITE was not sent					
	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of a RSC message: no action is required on the SIP side other than to terminate local procedures if any are in progress.					
SIP Parameter						
values:						
ISUP Parameter values:						
Comments:	ISUP/BICC SUT	SIP				
	IAM →					
	RSC →					
	RLC ←					

TP310002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit	: message (RSC)
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:		with the complete called party number, message before a SIP MESSAGE_VA until a SIP response has been received.
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP/BICC SUT IAM → RSC → RLC ←	SIP → INVITE ← SIP MESSAGE_VA → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK

Table 19

	Values for test purpose TP310002			
VA	VA SIP MESSAGE_VA			
VA_1	100 Trying			
VA_2	180 Ringing			
VA_3	181 Call Is Being Forwarded			
VA_4	182 Queued			
VA_5	183 Session Progress			

TP310003	SIP reference:	RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Re	eceipt of Reset circuit n	nessage (RSC)
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	 sending a INVITE mess has been received: On subsequer ACK for the 20 has been sent A Reason hear 	ter receiving the IAM wasage on receipt RSC manual receiving 200 OK INDO OK INVITE and substitute the rield containing the	OK INVITE is received ith the complete called party number, essage before a 200 OK response message NVITE messages, the SUT shall send an sequently send a BYE request after the ACK (ITU-T Recommendation Q.850 [5]) Cause age to be sent by the SIP side of the
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC	SUT	SIP
	IAM	→	→ INVITE
			← 100 Trying
	RSC	→	→ CANCEL
	RLC	←	◆ 200 OK INVITE
			→ ACK
			← 200 OK CANCEL
			→ BYE
		<u> </u>	← 200 OK BYE

TP310004	SIP reference	: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ F	Receipt of Reset circuit	message (RSC)
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:			VITE was sent. A BYE is sent
		•	with the complete called party number,
			e called party number, sending a BYE
	message on receipt R	SC message after a 20	00 OK response message has been received:
	TI 011T 1 11	L D)/E	
		nd a BYE request.	
	A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.		
SIP Parameter		<u> </u>	,
values:			
ISUP Parameter			
values:			
Comments:	ISUP/BICC	SUT	SIP
	IAM	→	→ INVITE
	ACM	←	← 180 Ringing
	ANM	←	← 200 OK INVITE
			→ ACK
	RSC	→	→ BYE
	RLC		← 200 OK BYE

TP310005	SIP reference: RFC	3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receip	ot of Reset circuit m	nessage (RSC)
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	 RSC in early dialogue received. A CANCEL is sent Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: The SUT shall send a CANCEL or BYE request. A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC IAM →	SUT	SIP → INVITE ← SIP MESSAGE_VA
	RSC → RLC ←		 → CANCEL ← 200 OK CANCEL ← 487 Request terminated
			→ ACK

Table 20

	Values for test purpose; TP310005			
VA	VA SIP MESSAGE_VA			
VA_1	180 Ringing			
VA_2	181 Call Is Being Forwarded			
VA_3	182 Queued			
VA_4	183 Session Progress			

6.2.2.10.2 Receipt of Circuit group reset message (GRS)

TP311001	SIP reference: RFC 3261 [6	5]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receipt of C	ircuit gr	oup reset message (GRS)
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	GRS received while an INVITE was not sent Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of GRS message: • no action is required on the SIP side other than to terminate local procedures if any are in progress.		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC S	UT	SIP
	IAM →		
	GRS →		
	GRA ←		

TP311002	SIP reference: RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)		
SIP selection criteria:			
ISUP selection			
criteria:			
Test purpose:	The SUT shall send a CANCA Reason header field conta	ne IAM with the t GRS message: message untice t request. In the time time the time time time time time time time tim	e complete called party number,
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC SI	JT	SIP
	IAM →	→	INVITE
	GRS →		
	GRA ←	←	SIP MESSAGE_VA
		→	OTHIOLL
		+	200 OK CANCEL
		+	487 Request terminated
		<u>→</u>	ACK

Table 21

Values for test purpose TP311002			
VA	A SIP MESSAGE_VA		
VA_1	100 Trying		
VA_2	180 Ringing		
VA_3	181 Call Is Being Forwarded		
VA_4	182 Queued		
VA 5	183 Session Progress		

TP311003	SIP reference: RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit	group reset	message (GRS)
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	 has been received: The SUT shall hold the GRS mess CANCEL is sent. On subsequently receiving 200 Ol 	e IAM with th GRS messa sage until a r	e complete called party number, ge before a 200 OK response message
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC SU	T	SIP
	IAM →	←	INVITE 100 Trying CANCEL
	GRS → GRA ←	← → ←	200 OK INVITE ACK 200 OK CANCEL BYE
			200 OK BYE

TP311004	SIP reference: RFC 3	261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receipt of	of Circuit group re	eset message (GRS)
SIP selection	·		
criteria:			
ISUP selection criteria:			
Test purpose:	GRS received after the ACK for	or a 200 OK INIVI	TE was sent A RVE is sent
rest purpose.	Ensure that the SUT after recessending a INVITE message wis message on receipt GRS mes The SUT shall send a BY A Reason header field containing the sum of the sum o	eiving the IAM with the complete of sage after a 200 E request.	th the complete called party number, called party number, sending a BYE OK response message has been received: -T Recommendation Q.850 [5]) Cause to be sent by the SIP side of the
SIP Parameter values:			
ISUP Parameter			
values:			
Comments:	ISUP/BICC	SUT	SIP
	IAM →		→ INVITE
	ACM ←		← 180 Ringing
	ANM ←		← 200 OK INVITE
			→ ACK
	GRS →		→ BYE
	GRA ←		← 200 OK BYE

TP311005	SIP reference: RFC 32	61 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receipt of	Circuit group	reset message (GRS)
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	GRS in early dialogue received		
			with the complete called party number,
			nessage after an early dialogue with the SIP
	message defined with the SIP_	MESSAGE_V	A has been established:
	The OUT of all count of OAA	1051	
	The SUT shall send a CAN		
			U-T Recommendation Q.850 [5]) Cause
SIP Parameter	value # 31 is added to the	SIP message	to be sent by the SIP side of the O-MGCF.
values:			
ISUP Parameter			
values:			
Comments:	ISUP/BICC	SUT	SIP
Comments.	IAM →	301	→ INVITE
	I/AlVI		SIP MESSAGE VA
	GRS →		V SII WESSAGE_VA
	GRA +		
	GRA		→ CANCEL
			◆ 200 OK CANCEL
			- ioi itoquoot toiiimatou
	1		→ ACK

Table 22

	Values for test purpose TP309009; TP311005				
VA	VA SIP MESSAGE_VA				
VA_1	180 Ringing				
VA_2	181 Call Is Being Forwarded				
VA_3	182 Queued				
VA 4	183 Session Progress				

TP311006	SIP reference: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)		
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	GRS for more than one CIC received. Send		
	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a GRS message were the Range Parameter value is bigger than "1": the SUT shall send a BYE requests for each call association.		
SIP Parameter	The controlled a bit is a square in the	Ouer su.	43300141011.
values:			
ISUP Parameter			
values:			
Comments:	ISUP/BICC SUT		SIP
	IAM(1) →	→	INVITE(1)
	ACM ←	←	180 Ringing
	ANM ←	←	200 OK INVITE
		→	ACK
	IAM(2) →	→	INVITE(2)
	ACM ←	←	180 Ringing
	ANM ←	←	200 OK INVITE
		→	ACK
	GRS(1) →	→	BYE(1)
	GRA ←	←	• •
		→	
		←	200 OK BYE
NOTE: BYE(1) aı	nd BYTE(2) possible received in reverse order	er.	

6.2.2.10.3 Receipt of Circuit group blocking message (CGB)

TP312001	SIP reference	e: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ hardware failure orier		blocking message (CGB) with the indication
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	CGB received while an INVITE was not sent Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented": no action is required on the SIP side other than to terminate local procedures if any are in progress.		
SIP Parameter values:			
ISUP Parameter values:	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented"		
Comments:	ISUP/BICC IAM CBG CGBA	SUT → ←	SIP

TP312002	SIP reference: RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.2.15	
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit hardware failure oriented	group blocki	ing message (CGB) with the indication	
SIP selection	naraware failure oriented			
criteria:				
ISUP selection				
criteria:				
Test purpose:	CGB received while no response for ar			
	Ensure that the SUT after receiving the			
			ge Circuit Group Supervision Message	
	Type Indicator coded as "hardware faile		before a SIP MESSAGE_VA	
	response message has been received:			
	The CLIT shall hald the CCD mass	ogo ustil o C	SID 200 OK reepense has been	
	 The SUT shall hold the CGB mess received. 	age unill a s	SIP 200 OK response has been	
	100011001			
	The SUT shall send a CANCEL request. A Passan hander field containing the (ITLL T Passanmendation O 850 [5]) Course.			
	A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.			
SIP Parameter	Talab ii 5 : 15 added to the on introduge to be contrally the on lottle of the original			
values:				
ISUP Parameter	CGB/CGBA: Circuit Group Supervision	Message T	ype Indicator coded as "hardware	
values:	failure oriented"			
Comments:	ISUP/BICC SU	Т	SIP	
	IAM →	→	INVITE	
	CGB →			
	CGBA ←	←	SIP MESSAGE_VA	
		→	CANCEL	
		←	200 OK CANCEL	
		←	487 Request terminated	
	<u> </u>	→	ACK	

Table 23

Values for test purpose TP312002			
VA	VA SIP MESSAGE_VA		
VA_1	100 Trying		
VA_2	180 Ringing		
VA_3	181 Call Is Being Forwarded		
VA_4	182 Queued		
VA_5	183 Session Progress		

TP312003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15		
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group hardware failure oriented	blocking message (CGB) with the indication		
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	Ensure that the SUT after receiving the IAM sending a INVITE message on receipt CGB Type Indicator coded as "hardware failure or has been received: On subsequently receiving 200 OK INVI the 200 OK INVITE and subsequently sent A Reason header field containing the (IT)	While CANCEL is sent, a 200 OK INVITE is received SUT after receiving the IAM with the complete called party number, TE message on receipt CGB message Circuit Group Supervision Message coded as "hardware failure oriented" before a 200 OK response message red: Luently receiving 200 OK INVITE messages, the SUT shall send an ACK for INVITE and subsequently send a BYE request after the ACK has been Leader field containing the (ITU-T Recommendation Q.850 [5]) Cause is added to the SIP message to be sent by the SIP side of the O-MGCF.		
SIP Parameter values:				
ISUP Parameter	CGB/CGBA: Circuit Group Supervision Mes	sage Type Indicator coded as "hardware		
values: Comments:	failure oriented" ISUP/BICC SUT	SIP		
Comments.	IAM →	→ INVITE ← 100 Trying		
	CGB → CGBA ←	→ CANCEL ← 200 OK INVITE → ACK ← 200 OK CANCEL → BYE ← 200 OK BYE		

TP312004	SIP reference: RFC 3261 [6]	ISUP reference:				
11 312004	on reference. Ki o 3201 [o]	ES 283 027 [1], clause 7.2.3.2.15				
TSS reference:	ICLID CID/Dagio call/ Daggint of Circuit group	P/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication				
133 reference:	hardware failure oriented	blocking message (CGB) with the indication				
SIP selection	nardware failure oriented					
criteria:						
ISUP selection						
criteria:						
Test purpose:	CGB received after the ACK for a 200 OK IN	VITE was sent. A RYE is sent				
Tool parpood.	Ensure that the SUT after receiving the IAM					
	sending a INVITE message with the complete					
	message on receipt CGB message Circuit G					
	coded as "hardware failure oriented" after a					
	received:	-oo on roopense meesage nas zeen				
	 The SUT shall send a BYE request. 					
	·	U-T Rec Q.850 [5]) Cause Value # 31 is				
	added to the SIP message to be sent by					
SIP Parameter	added to the Oil Message to be sont by the Oil Side of the Oil Mesor.					
values:						
ISUP Parameter	CGB/CGBA: Circuit Group Supervision Mess	age Type Indicator coded as "hardware				
values:	failure oriented"					
Comments:	ISUP/BICC SUT	SIP				
	IAM →	→ INVITE				
	ACM ←	← 180 Ringing				
	ANM ←	← 200 OK INVITE				
		→ ACK				
	CGB →	→ BYE				
	CGBA ←	← 200 OK BYE				

TP312005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.15		
TSS reference:	1	up blocking message (CGB) with the indication		
OID I I	hardware failure oriented			
SIP selection				
criteria:				
ISUP selection criteria:				
Test purpose:	sending a INVITE message on receipt CGI Type Indicator coded as "hardware failure message defined with the SIP_MESSAGE The SUT shall send a CANCEL reque A Reason header field containing the	after receiving the IAM with the complete called party number, lessage on receipt CGB message Circuit Group Supervision Message d as "hardware failure oriented" after an early dialogue with the SIP th the SIP_MESSAGE_VA has been established:		
values:				
ISUP Parameter	CGB/CGBA: Circuit Group Supervision Me	ssage Type Indicator coded as "hardware		
values:	failure oriented"	obago Typo maioator obaba ao marawaro		
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
		← SIP MESSAGE_VA		
	CGB →			
	CGBA ←			
		→ CANCEL		
		← 200 OK CANCEL		
		← 487 Request terminated		
		→ ACK		

Table 24

Values for test purpose ; TP312005			
VA	VA SIP MESSAGE_VA		
VA_1	180 Ringing		
VA_2	181 Call Is Being Forwarded		
VA_3	182 Queued		
VA_4	183 Session Progress		

TP312006	SIP reference	ce: RFC 3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.2.3.2.15
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	CGB for more than one CIC received. Send a BYE for each circuit Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" where the Range and Status Parameter value is bigger than "1": • the SUT shall send a BYE requests for each call association.			
SIP Parameter values:				
ISUP Parameter values:	CGB/CGBA: Circuit failure oriented"	Group Supervision Messa	age Type	Indicator coded as "hardware
omments:	ISUP/BICC IAM(1) ACM ANM	SUT → ← ←	→ ← ←	SIP INVITE(1) 180 Ringing 200 OK INVITE ACK
	IAM(2) ACM ANM	→ ← ←	→ ← ←	INVITE(2) 180 Ringing 200 OK INVITE ACK
CGB(1) CGBA BYE(1) CGBA CGBA BYE(2) COBA BYE(2) COBA BYE(1) BYE(2) COBA BYE(2) COBA C			200 OK BYE BYE(2)	

6.3.1 Interworking from SIP to ISUP (Incoming Call)

6.3.1.1 Calling Line Identification (CLI)

TP501001	SIP reference: RFC 3261 [6]	ISUP reference:		
		ES 283 027 [1], clause 7.	2.3.1.2.6	
	SIP-ISUP/SS/CLI/			
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	P-Asserted-Identity not in E.164 format and From header not in the E.164 Format, no Privacy header. Send Calling party number			
	Ensure that the SUT in the Idle state, on recei	_		
	 the SIP P-Asserted-Identity containing NDC+ SN has not been received; 	•		
	 the SIP From header field containing a NDC+ SN has not been received; 	•	nat "+" CC+	
	a Privacy header field has not been r	eceived.		
	sends an IAM message with the Calling party number parameter coded:			
	Address signals = absent			
	Screening indicator = network provided			
	Number Incomplete Indicator = incomplete			
	Numbering plan indicator = 000			
	Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000			
SIP Parameter	Nature of address indicator = 0000000			
values:				
ISUP Parameter				
values:				
Comments:	SIP SU	T ISUP		
	INVITE →	→ IAM		
	180 Ringing ←	← ACM		
	Ringing			
	200 OK INVITE	← ANM		
	ACK →			
	Conver	sation		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

Table 25: Values for test purposes TP501001

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501002	SIP reference: RFC 3261 [6]		SUP reference:
			27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection			
criteria:	D.A	- , ,	
Test purpose:	P-Asserted-Identity not in E.164 format and F value none. Send Calling party number Ensure that the SUT in the Idle state, on rece the SIP P-Asserted-Identity containing CC+ NDC+ SN has not been received; the SIP From header field containing NDC+ SN has not been received; a Privacy header field was received "none". sends an IAM message with the Calling part Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplet Numbering plan indicator = 000 Address Presentation Restricted Indicator Nature of address indicator = 00000000	ipt of a INVITE ong a URI with an wed; g a URI with an in and the priv-valuey number parage	message where: n identity in the format "+" identity in the format "+" CC+ ue component is set to
SIP Parameter			
values:			
ISUP Parameter			
values: Comments:	SIP S	UT	ISUP
Comments.	INVITE →	∪ı →	IAM
	180 Ringing	7	ACM
		ng tone	ACIVI
	200 OK INVITE ←	ig torie	ANM
	ACK →	•	, vi Aivi
		ersation	
	BYE →	→	REL
	200 OK BYE ←	É	RLC

TP501003	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/	E3 203 U	27 [1], Clause 7.2.3.1.2.0
SIP selection	31F-130F/33/CLI/		
criteria:			
ISUP selection			
criteria:			
Test purpose:	P-Asserted-Identity not in E. 164 format and Fivalue header. Send Calling party number Ensure that the SUT in the Idle state, on received: • the SIP P-Asserted-Identity containing CC+ NDC+ SN has not been received: • the SIP From header field containing NDC+ SN has not been received: • a Privacy header field was received "header". sends an IAM message with the Calling part Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator Nature of address indicator = 00000000	ipt of a INVITE ng a URI with ar wed; g a URI with an and the priv-val y number para	message where: n identity in the format "+" identity in the format "+" CC+ ue component is set to
SIP Parameter			
values: ISUP Parameter			
values:			
Comments:	SIP S	JT	ISUP
	INVITE →	→	IAM
	180 Ringing ←	←	ACM
	•	g tone	
	200 OK INVITE ←	←	ANM
	ACK →		
		rsation	
	BYE →	→	REL
	200 OK BYE ←	+	RLC

TP501004	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/	E3 203 U2	27 [1], clause 7.2.3.1.2.0
SIP selection	51P-15UP/55/CLI/		
criteria:			
ISUP selection			
criteria:			
	P-Asserted-Identity not in E 164 format and E	rom header not	in the F 164 Format Privacy
Test purpose:	P-Asserted-Identity not in E.164 format and Fivalue user. Send Calling party number Ensure that the SUT in the Idle state, on recei the SIP P-Asserted-Identity containing NDC+ SN has not been received; the SIP From header field containing NDC+ SN has not been received; a Privacy header field was received as sends an IAM message with the Calling party Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator	pt of a INVITE r g a URI with an id a URI with an id nd the priv-value number parar	nessage where: identity in the format "+" CC+ entity in the format "+" CC+ e component is set to "user".
OID Danamatan	Nature of address indicator = 0000000		
SIP Parameter values:			
ISUP Parameter			
values:			
Comments:	SIP SU	IT	ISUP
	INVITE →	·· →	IAM
	180 Ringing	÷	ACM
	Ringing	=	/ (OIVI
	200 OK INVITE	y torre	ANM
	ACK →	~	/ VI AIAI
	Conver	reation	
	BYE -	Salion +	REL
	200 OK BYE ←	→	RLC
	ZUU UN DIE		RLU

TP501005	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/	1	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	P-Asserted-Identity not in E.164 format and F value id. Send Calling party number Ensure that the SUT in the Idle state, on rece the SIP P-Asserted-Identity containin NDC+ SN has not been received; the SIP From header field containing NDC+ SN has not been received; a Privacy header field was received a sends an IAM message with the Calling part Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator Nature of address indicator = 00000000	ipt of a INVITE g a URI with an a URI with an io and the priv-valu y number para	message where: identity in the format "+" CC+ dentity in the format "+" CC+ le component is set to "id".
SIP Parameter			
values:			
ISUP Parameter values:			
Comments:	SIP SI	JT	ISUP
	INVITE →	→	IAM
	180 Ringing ←	-	ACM
		g tone	-
	200 OK INVITE	←	ANM
	ACK →		
	Conve	rsation	
	BYE →	→	REL
	200 OK BYE ←	+	RLC

TP501006	SIP reference: RFC 3261 [6]	ISUP reference:		
		ES 283 027 [1], clause 7.2.3.1.2.6		
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection criteria:				
ISUP selection criteria:	PICS 6/3			
Test purpose:	 P-Asserted-Identity not in E.164 format and From header in the E.164 Format, no Privacy header received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; a Privacy header field has not been received. 			
	sends an IAM message with the Calling part Address signals = absent	y number parameter coded:		
	Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000			
CID Daysmeter	with the Generic number parameter coded: Address signals = derrived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	SIP SI	JT ISUP		
2 3	INVITE →	→ IAM		
	180 Ringing ←	← ACM		
		ig tone		
	200 OK INVITE	← ANM		
	ACK →			
		rsation		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

Table 26: Values for test purposes TP501006

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501007	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/	L3 203 0	21 [1], clause 1.2.5.1.2.0	
SIP selection	011 1001 700/021/			
criteria:				
ISUP selection	PICS 6/3			
criteria:				
Test purpose:	P-Asserted-Identity not in E.164 format and From header in the E.164 Format, Privacy value none received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where:			
	the SIP P-Asserted-Identity contain CC+ NDC+ SN has not been reco	eived;	•	
	 the SIP From header field containi NDC+ SN has been received; 		•	
	a Privacy header field was receive "none".	d and the priv-val	ue component is set to	
	sends an IAM message with the Calling pa	rty number para	meter coded:	
	Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000			
	with the Generic number parameter coded:			
	Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	SIP	SUT	ISUP	
	INVITE →	→	IAM	
	180 Ringing ←	.	ACM	
		ing tone	A N I N A	
	200 OK INVITE ← ACK →	←	ANM	
	_	vorantian		
	BYE →	rersation	REL	
	200 OK BYE ←	7	RLC	
	ZUU UN DIE T		RLU	

Table 27: Values for test purposes TP501007

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501008	SIP reference: RFC 3261 [6]	I	SUP reference:	
		ES 283 0	27 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection criteria:				
ISUP selection criteria:	PICS 6/3			
Test purpose:	P-Asserted-Identity not in E.164 format and From header in the E.164 Format, Privacy value header received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where:			
	 the SIP P-Asserted-Identity contain CC+ NDC+ SN has not been rece the SIP From header field containing 	ved;	·	
	 NDC+ SN has been received; a Privacy header field was received "header". 	and the priv-val	ue component is set to	
	sends an IAM message with the Calling par	ty number para	meter coded:	
	Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000			
	with the Generic number parameter coded			
	Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA VALUE			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	SIP S	UT	ISUP	
	INVITE →	→	IAM	
	180 Ringing ←	←	ACM	
	_	ng tone		
	200 OK INVITE ←	←	ANM	
	ACK →			
		ersation		
	BYE →	→	REL	
	200 OK BYE ←	<u> </u>	RLC	

Table 28: Values for test purposes TP501008

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501009	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/	•		
SIP selection criteria:				
ISUP selection criteria:	PICS 6/3			
Test purpose:	P-Asserted-Identity not in E.164 format and From header in the E.164 Format, Privacy value user received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "user".			
	sends an IAM message with the Calling p Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomp Numbering plan indicator = 000 Address Presentation Restricted Indica Nature of address indicator = 0000000 with the Generic number parameter code Address signals = number provided by Screening indicator = user provided, no Number Incomplete Indicator = comple Numbering plan indicator = ISDN numb Address Presentation Restricted Indicator NoAS: NoA_VALUE	lete tor = PIXIT d: the user of verified te pering plan		
SIP Parameter				
values:				
values:				
Comments:	SIP	SUT	ISUP	
Comments.	INVITE →	→	IAM	
	180 Ringing ←	É	ACM	
	100 1 11191119	ging tone	7.0101	
	200 OK INVITE ←	ging tone	ANM	
	ACK →	•	, 77 41A1	
		versation		
	BYE →	→	REL	
	200 OK BYE	-	RLC	
Ĺ	ZUU UN DIE T		NLU	

Table 29: Values for test purposes TP501009

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501010	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/3		
Test purpose:	P-Asserted-Identity not in E.164 format and From header in the E.164 Format, Privacy value id received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "id". sends an IAM message with the Calling party number parameter coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT		
	Nature of address indicator = 0000000 with the Generic number parameter coded: Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP SI	UT ISUP	
	INVITE ->	→ IAM	
	180 Ringing ←	← ACM	
		ng tone	
	200 OK INVITE	← ANM	
	ACK →	- / 11 1111	
		ersation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	
<u> </u>	ZUU UN DIL T	₹ INLO	

Table 30: Values for test purposes TP501010

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501011	SIP reference: RFC 3261 [6]	l	SUP reference:	
		ES 283 0	27 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection				
criteria:				
ISUP selection				
criteria:	D.A		L E 101 E	
Test purpose:	P-Asserted-Identity in E.164 format and From	neader not in t	ne E.164 Format, no Privacy	
	header received. Send Calling party number Ensure that the SUT in the Idle state, on rece	int of a INIVITE	mossago whoro:	
	Linsule that the 501 in the lide state, on rece	ipt of a liveric	message where.	
	the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received;			
	the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;			
	a Privacy header field has not been received.			
	sends an IAM message with the Calling party number parameter coded:			
	Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE			
SIP Parameter				
values:				
ISUP Parameter				
values: Comments:	SIP SI	JT	ISUP	
Comments.	INVITE →	- →	IAM	
	180 Ringing	-	ACM	
	100 111191119	g tone	/ CIVI	
	200 OK INVITE	←	ANM	
	ACK →	-		
	7.6.1	rsation		
	BYE →	→	REL	
	200 OK BYE ←	+	RLC	

Table 31: Values for test purposes TP501011

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501012	SIP reference: RFC 3261 [6]	ISUP reference:		
<i>(</i>		ES 283 027 [1], clause 7.2.3.1.2.6		
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection				
criteria:				
ISUP selection				
criteria:	D. Asserted Identity in E 164 format and From	hander not in the F 164 Format Drivery		
Test purpose:	P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy			
	value none received. Send Calling party numbers in the the SUT in the Idle state, on received.			
	Lisure that the SOT in the late state, on recei	pt of a INVITE message where.		
	the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received;			
	 the SIP From header field containing a NDC+ SN has not been received; 	a URI with an identity in the format "+" CC+		
	a Privacy header field was received and the priv-value component is set to "none".			
	sends an IAM message with the Calling party number parameter coded:			
	Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE			
SIP Parameter				
values:				
ISUP Parameter				
values:				
Comments:	SIP SU			
	INVITE →	→ IAM		
	180 Ringing ←	← ACM		
	Ringin			
	200 OK INVITE ←	← ANM		
	ACK →			
	Conver			
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

Table 32: Values for test purposes TP501012

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA 01	a different country	"International number"	CC+NDC+SN

TP501013	SIP reference: RFC 3261 [6]	_	SUP reference: 27 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	 P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value header received. Send Calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; a Privacy header field was received and the priv-value component is set to "header". 			
	sends an IAM message with the Calling party number parameter coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted			
SIP Parameter				
values:				
ISUP Parameter values:				
Comments:	SIP SU	JT	ISUP	
	INVITE →	→	IAM	
	180 Ringing	+	ACM	
	Ringin	g tone		
	200 OK INVITE ←	+	ANM	
	ACK →			
	Conve	rsation		
	BYE →	→	REL	
	200 OK BYE ←	←	RLC	

TP501014	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	 P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value user received. Send Calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; a Privacy header field was received and the priv-value component is set to "user". sends an IAM message with the Calling party number parameter coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted 		
SIP Parameter			
values: ISUP Parameter			
values:			
Comments:	SIP SU	T	ISUP
Joninionts.	INVITE →	'. →	IAM
	180 Ringing	÷	ACM
	Ringing	_	7.0
	200 OK INVITE	←	ANM
	ACK →	-	
	Conver	sation	
	BYE →	→	REL
	200 OK BYE ←	<u></u>	RLC

TP501015	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value id received. Send Calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field was received and the priv-value component is set to "id". sends an IAM message with the Calling party number parameter coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted		
SIP Parameter			
values:			
ISUP Parameter			
values:			
Comments:	SIP		ISUP
	INVITE -	→	IAM
	180 Ringing ←	+	ACM
	Ringin	•	
	200 OK INVITE	+	ANM
	ACK →		
	Conve		
	BYE →	→	REL
	200 OK BYE ←	<u> </u>	RLC

TP501016	SIP reference: RFC 3261 [6]	ISUP reference:			
11 301013	5.1 Telefelioe: 10 0 0201 [0]	ES 283 027 [1], clause 7.2.3.1.2.6			
TSS reference:	SIP-ISUP/SS/CLI/	h 4/			
SIP selection					
criteria:					
ISUP selection	PICS 6/3				
criteria:					
Test purpose:	P-Asserted-Identity in E.164 format and From				
	header received. Send Calling party number a				
	Ensure that the SOT in the late state, on rece	Ensure that the SUT in the Idle state, on receipt of a INVITE message where:			
	 the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; the SIP From header field containing a URI with an identity in the format "+" CC+ 				
	NDC+ SN has been received;a Privacy header field has not been re	eceived.			
	sends an IAM message with the Calling part Address signals = number derived from S Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numberi Address Presentation Restricted Indicator NoAS: NoA_VALUE with the Generic number parameter coded: Address signals = number derived from the Screening indicator = user provided, not vent in Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering Address Presentation Restricted Indicator NoAS: NoA_VALUE	ing plan r = Presentation allowed the From header verified ing plan			
SIP Parameter					
values:					
ISUP Parameter					
values: Comments:	SIP SI	UT ISUP			
Comments:	INVITE →	OT ISOP → IAM			
	180 Ringing	→ IAIVI ← ACM			
		ng tone			
	200 OK INVITE	← ANM			
	ACK →				
		ersation			
	BYE →	→ REL			
	200 OK BYE ←	← RLC			

Table 33: Values for test purposes TP501016

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501017	SIP reference: RFC 3261 [6]	ISUP reference:	
11 301017	Sir Telefence. Ki C 3201 [0]	ES 283 027 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection	PICS 6/3		
criteria:			
Test purpose:	P-Asserted-Identity in E.164 format and From header in the E.164 Format, Privacy value none received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to		
	"none". sends an IAM message with the Calling part Address signals = number derived from S Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN number Address Presentation Restricted Indicator NoAS: NoA_VALUE with the Generic number parameter coded: Address signals = number derived from t Screening indicator = user provided, not to Number Incomplete Indicator = complete Numbering plan indicator = ISDN number Address Presentation Restricted Indicator NoAS: NoA_VALUE	SIP P-Asserted-Identity ring plan r = Presentation allowed the From header verified ring plan	
SIP Parameter			
values:			
ISUP Parameter values:			
Comments:	SIP S	UT ISUP	
Joninion S.	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		ng tone	
	200 OK INVITE	← ANM	
	ACK →		
		ersation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

Table 34: Values for test purposes TP501017

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501018	SIP reference: RFC 3261 [6]	I	SUP reference:
		ES 283 02	27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/3		
Test purpose:	P-Asserted-Identity in E.164 format and From header in the E.164 Format, Privacy value header received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "header".		
	sends an IAM message with the Calling par Address signals = number derived from S Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN number Address Presentation Restricted Indicato NoAS: NoA_VALUE with the Generic number parameter coded: Address signals = number derived from t Screening indicator = user provided, not Number Incomplete Indicator = complete Numbering plan indicator = ISDN number Address Presentation Restricted Indicato NoAS: NoA_VALUE	ing plan = Presentation he From header erified ing plan	dentity
SIP Parameter values:			
ISUP Parameter			
values:			
Comments:	SIP	UT	ISUP
	INVITE →	→	IAM
	180 Ringing ←	←	ACM
	Ringii	ig tone	
	200 OK INVITE ←	←	ANM
	ACK →		
	Conve	rsation	
	BYE →	→	REL
	200 OK BYE ←	+	RLC

Table 35: Values for test purposes TP501018

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

TP501019	SIP reference: RFC 3261 [6]		P reference:
		ES 283 027 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/3		
Test purpose:	 P-Asserted-Identity in E.164 format and From header in the E.164 Format, Privacy value user received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; a Privacy header field was received and the priv-value component is set to "user". sends an IAM message with the Calling party number parameter coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE with the Generic number parameter coded: Address signals = number derived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Number Incomplete Indicator = ISDN numbering plan 		
SIP Parameter	NoAS: NoA_VALUE		
values:			
ISUP Parameter values:			
Comments:	SIP	IT IS	SUP
	INVITE →		AM
	180 Ringing ←	← A	CM
	Ringin	g tone	
	200 OK INVITE ←	← A	NM
	ACK →		
	Conve	sation	
	BYE →		EL
	200 OK BYE ←	← R	LC

Table 36: Values for test purposes TP501019

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

3.1.2.6		
P-Asserted-Identity in E.164 format and From header in the E.164 Format, Privacy value id received. Send Calling party number and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received;		
et to "id".		

TP501021	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/1		
Test purpose:	P-Asserted-Identity not in E.164 format and F Privacy header. Send Calling party number not Ensure that the SUT in the Idle state, on received: • the SIP P-Asserted-Identity containing NDC+ SN has not been received; • the SIP From header field containing NDC+ SN has not been received; • a Privacy header field has not been reserved and IAM message with the Calling part Address signals = network provided (PIXIS) Screening indicator = network provided Nature of address indicator = NoA_VALUE Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering Address Presentation Restricted Indicator	etwork provided ipt of a INVITE of a INVITE of a INVITE of a INVITE of a URI with an ideaceived. y number para of the control	message where: identity in the format "+" CC+ dentity in the format "+" CC+
SIP Parameter values:			
ISUP Parameter			
values: Comments:	SIP SU	IT.	ISUP
Comments:	INVITE →	JI →	IAM
		→	ACM
		=	ACIVI
	Ringin	g tone	ANINA
	200 OK INVITE ← ACK →	~	ANM
	7.011		
	Conve		DEL
	BYE 200 OK BYE ←	→	REL
	200 OK BYE ←		RLC

TP501022	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	P-Asserted-Identity sip URI, without user=phone and P-Asserted-Identity tel URI, no Privacy header. Send Calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a SIP URI with an identity 1 in the format "+" CC+ NDC+ SN has been received without user = phone; • the SIP P-Asserted-Identity containing a Tel URI with an identity 2 in the format "+" CC+ NDC+ SN has been received; • a Privacy header field has not been received. sends an IAM message with the Calling party number parameter coded: Address signals = identity 2 Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE		
SIP Parameter values:			
ISUP Parameter			
values:			
Comments:	SIP S	UT	ISUP
	INVITE ->	→	IAM
	180 Ringing ←	←	ACM
	Ringir	ng tone	
	200 OK INVITE ←	←	ANM
	ACK →		
	Conve	ersation	
	BYE →	→	REL
	200 OK BYE ←	+	RLC

TP501023	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection	PICS 6/1 AND PICS 6/12		
criteria:			
Test purpose:	P-Asserted-Identity not in E.164 format, no Privacy header. Send Calling party number network provided Address not available Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field has not been received. sends an IAM message with the Calling party number parameter coded: Address signals = not present Screening indicator = network provided Number Incomplete Indicator = incomplete Address Presentation Restricted Indicator = Address not available		
SIP Parameter	/taaroos i rosentation restricted maistator	- Madross Hot available	
values:			
ISUP Parameter			
values:			
Comments:		UT ISUP	
	INVITE -	→ IAM	
	180 Ringing ←	← ACM	
		ng tone	
	200 OK INVITE ←	← ANM	
	ACK →		
		ersation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP501024	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/1 AND PICS 6/3 AND PICS 6/12		
Test purpose:	P-Asserted-Identity not in E.164 format and From header in the E.164 Format, no Privacy header received. Send Calling party number network provided and Additional calling party number Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+		
	NDC+ SN has been received;a Privacy header field has not been	received.	
	sends an IAM message with the Calling par	ty number parameter coded:	
	Address signals = not present Screening indicator = network provided Number Incomplete Indicator = incomplete Address Presentation Restricted Indicator = Address not available		
CID Parameter	with the Generic number parameter coded: Address signals = number derived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP	SUT ISUP	
Comments.	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		ng tone	
	200 OK INVITE	← ANM	
	ACK →		
		ersation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

Table 37: Values for test purposes TP501122, TP501023, TP501024

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	'National (Significant) number"	NDC+SN
VA_01	a different country	"International number"	CC+NDC+SN

6.3.1.2 Call Hold (HOLD)

TP502001	SIP reference: RFC 3261 [6]	IS	UP reference:	
			7.4.10/[14]	
TSS reference:	SIP-ISUP/SS/HOLD/			
SIP selection	PICS 8/4			
criteria:				
ISUP selection	PICS 5/22			
criteria:				
Test purpose:	Each party can hold and retrieve the remote p	Each party can hold and retrieve the remote party in the confirmed state		
	Ensure that a party can put the other party on and before call clearing has begun. Ensure the on hold.			
	The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party The called party should be able to put the other party on hold The called party should be able to retrieve the other party			
SIP Parameter	SDP: a=sendonly (put on hold)	other party		
values:	a=sendrecv or omitted (retrieve the call)		
Tuluo01	o= <version incremented=""></version>	,		
ISUP Parameter	CPG: Generic notification: remote hold Event	ndicator PROGF	RESS (put on hold)	
values:	Generic notification: remote retrieval eve	nt indicator PRO	GRESS (retrieve the call)	
Comments:	-	CF	ISUP	
	INVITE →	→	IAM	
	180 Ringing ←	(ACM	
	200 OK INVITE ←	(ANM	
	INVITE(sendonly) →	→	CPG(hold)	
	200 OK INVITE(recvonly) ←		,	
	INVITE(sendrecv) →	→	CPG(retrieve)	
	200 OK INVITE(sendrecv) ←			
	INVITE(sendonly)	←	CPG(hold)	
	200 OK INVITE(recvonly) →			
	INVITE(sendrecv)	(CPG(retrieve)	
	200 OK INVITE(sendrecv) →			

TP502002	SIP reference: RFC 3261 [6]		UP reference: 7.4.10/[14]
TSS reference:	SIP-ISUP/SS/HOLD/		
SIP selection criteria:	PICS 8/4		
ISUP selection criteria:	PICS 5/22 PICS 8/1		
Test purpose:	The calling party can hold and retrieve the remote party in the early dialogue Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold		
CID Donomotor	The calling party should be able to retrieve the	e otner party	
SIP Parameter values:	SDP: a=sendonly (put on hold)	11)	
values.	a=sendrecv or omitted (retrieve the ca o= <version incremented=""></version>	1)	
ISUP Parameter	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)		
values:	Generic notification: remote retrieval eve		
Comments:	SIP Me	GCF	ISUP
	INVITE ->	→	IAM
	180 Ringing ←	-	ACM
	UPDATE(sendonly) 200 OK UPDATE(recvonly) →	→	CPG(hold)
	UPDATE(sendrecv) → 200 OK UPDATE(sendrecv) ←	→	CPG(retrieve)

TP502003	SIP reference: RFC 3261 [6]	ISUP reference:	
		7.4.10/[14]	
TSS reference:	SIP-ISUP/SS/HOLD/		
SIP selection	PICS 8/2		
criteria:			
ISUP selection	PICS 5/22		
criteria:			
Test purpose:	The calling party can hold and retrieve the remote party after the calling party has provided all information to process the call		
	Ensure that a party can put the other party on hold after the calling user has provided all of the information necessary for processing the call. Ensure that the party can retrieve the call previously put on hold.		
	The calling party should be able to put the oth	er narty on hold	
	The calling party should be able to retrieve the		
SIP Parameter	SDP: a=sendonly (put on hold)		
values:	a=sendrecv or omitted (retrieve the call)		
	o= <version> incremented</version>		
ISUP Parameter	ACM: called party status: no indication		
values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)		
	Generic notification: remote retrieval Event indicator PROGRESS (retrieve the call)		
Comments:		GCF ISUP	
	INVITE →	→ IAM	
	UPDATE(sendonly) → 200 OK UPDATE(recyonly) ←		
	200 OK UPDATE(recvonly) ←		
	UPDATE(sendrecv) →		
	200 OK UPDATE(sendrecv) ←		

TP502004	SIP reference: RFC 326	1 [6]		UP reference: 7.4.10/[14]
TSS reference:	SIP-ISUP/SS/HOLD/			
SIP selection	PICS 8/4			
criteria:				
ISUP selection	PICS 5/22			
criteria:				
Test purpose:	A party can hold and retrieve the remote party in the confirmed state using the UPDATE method (receiving) Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put			
SIP Parameter	on hold. The calling party should be able The calling party should be able	to retrieve the		
values:	SDP: a=sendonly (put on hold)		١	
values.	a=sendrecv or omitted (retrieve the call) o= <version incremented=""></version>			
ISUP Parameter	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)			
values:	Generic notification: remot	e retrieval eve	nt indicator PRO	
Comments:	SIP		CF	ISUP
		→	→	IAM
	3 3	((ACM
	200 OK INVITE	←	(ANM
	UPDATE(sendonly)	→	→	CPG(hold)
	200 OK INVITE(recvonly)	(•
	UPDATE(sendrecv) 200 OK UPDATE(recvonly)	→	→	CPG(retrieve)

TP502005	SIP reference: RFC 3261 [6]	IS	SUP reference: 7.4.10/[14]
TSS reference:	SIP-ISUP/SS/HOLD/		
SIP selection	PICS 8/4 PICS 8/3		
criteria: ISUP selection	DICC 5/00		
criteria:	PICS 5/22		
Test purpose:	A party can hold and retrieve the remote party in the confirmed state using the UPDATE method (sending) Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The called party should be able to put the other party on hold The called party should be able to retrieve the other party		
SIP Parameter	SDP: a=sendonly (put on hold)	and daner pairty	
values:	a=sendrecv or omitted (retrieve the	call)	
	o= <version incremented=""></version>		
ISUP Parameter	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)		
values:	Generic notification: remote retrieva		
Comments:	SIP	MGCF	ISUP
	INVITE →	→	IAM
	180 Ringing ←	(ACM
	200 OK INVITE ←	←	ANM
	UPDATE(sendonly) 200 OK INVITE(recvonly) →	(CPG(hold)
	UPDATE(sendrecv) ← 200 OK UPDATE(recvonly) →	+	CPG(retrieve)

TP502006	SIP reference: RFC 3	3261 [6]	IS	UP reference: 7.4.10/[14]
TSS reference:	SIP-ISUP/SS/HOLD/			
SIP selection criteria:	PICS 8/4			
ISUP selection criteria:	PICS 5/22			
Test purpose:	Both parties can hold and retrieve the remote party in the confirmed state. First hold party retrieves first Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party in held state can put the remote party put on hold. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party.			
	The called party should be ab			
SIP Parameter values:		SDP: a=sendonly or a=inactive (put on hold) a=sendrecv or a=recvonly or omitted (retrieve the call)		
ISUP Parameter	CPG: Generic notification: rer	note hold Even	t indicator PROGE	RESS (put on hold)
values:	Generic notification: rem	note retrieval ev	ent indicator PRC	OGRESS (retrieve the call)
Comments:	SIP INVITE 180 Ringing 200 OK INVITE	→ ← ←	IGCF → ←	ISUP
	INVITE(sendonly) 200 OK INVITE(recvonly) INVITE(inactive)	→ ←	→	CPG(hold) CPG(hold)
	200 OK INVITE(inactive) INVITE(recvonly)	→	→	CPG(retrieve)
	200 OK INVITE(sendonly)	-	_	,
	INVITE(sendrecv) 200 OK INVITE(sendrecv)	← →	+	CPG(retrieve)

TP502007	SIP reference: RFC 32	61 [6]	IS	SUP reference: 7.4.10/[14]
TSS reference:	SIP-ISUP/SS/HOLD/			
SIP selection criteria:	PICS 8/4			
ISUP selection	PICS 5/22			
criteria:				
Test purpose:	Both partys can hold and retrieve the remote party in the confirmed state. Second hold party retrieves first Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party in held state can put the remote party put on hold. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The called party should be able to retrieve the other party			
	The calling party should be able			
SIP Parameter values:	SDP: a=sendonly or a=inactive (put on hold) a=sendrecv or a=recvonly or omitted (retrieve the call) o= <version incremented=""></version>			
ISUP Parameter	CPG: Generic notification: reme	ote hold Event	indicator PROGI	RESS (put on hold)
values:	Generic notification: remo	ote retrieval eve	ent indicator PRO	OGRESS (retrieve the call)
Comments:	SIP INVITE 180 Ringing 200 OK INVITE	MC → ←	GCF → ← ←	ISUP
	INVITE(sendonly) 200 OK INVITE(recvonly)	→	→	CPG(hold)
	INVITE(inactive) ← CPG(hold) 200 OK INVITE(inactive) →			
	INVITE(recvonly) 200 OK INVITE(sendonly)	← →	+	CPG(retrieve)
	INVITE(sendrecv) 200 OK INVITE(sendrecv)	→	→	CPG(retrieve)

6.3.1.3 Terminal portability (TP)

Void.

6.3.1.4 Conference calling (CONF)

TP504001	SIP reference: RFC 3261 [6]	ГС	NGN reference:
TCC vofevence:	OLD TOTAL CONTENT	E3	283 027 [1], clause 7.4.14
TSS reference:	SIP-ISUP/SS/CONF/		
SIP selection criteria:	PICS 8/2		
ISUP selection criteria:	PICS 5/10		
Test purpose:	Generic notification Conference established and Conference disconnected and SIP procedure		
	Ensure that the SUT does not stop the temporare streams if a CPG message Generic notification supplementary service.		
SIP Parameter			
values:			
ISUP Parameter	CPG: Generic notification = Conference est	ablished	
values:	CPG: Generic notification = Conference dis-	connected	
Comments:	SIP SU	Τ	ISUP
	INVITE →	→	IAM
	180 Ringing ←	+	ACM
	Ringing tone		
	200 OK INVITE ←	+	ANM
	ACK →		
	Conve	ersation	
		+	CPG(Conference established)
	Conve	ersation	
		←	CPG(Conference disconnected)
	BYE →	→	REL
	200 OK BYE ←	+	RLC

TP504002	SIP reference: RFC 3261 [6]	NGN reference:		
		ES 283 027 [1], clause 7.4.14		
TSS reference:	SIP-ISUP/SS/CONF/			
SIP selection criteria:	PICS 8/2			
ISUP selection criteria:	PICS 5/10			
Test purpose:	Generic notification Isolated and Reattached and SIP procedure			
	Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the CONF supplementary service. If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute			
SIP Parameter values:	line a=sendonly/sendrecv or omitted attributed SDP: a=sendonly/sentrecv or a line is omitted sendonly/sentrecv or a line is omitted.			
ISUP Parameter values:	CPG: Generic notification = Conference esta CPG: Generic notification = GEN_NOT_VAL CPG: Generic notification = Conference disco	UE		
Comments:	SIP SUT			
	INVITE →	→ IAM		
	180 Ringing ←	← ACM		
	Ringing tone			
	200 OK INVITE ←	← ANM		
	ACK →			
	Conver			
		← CPG(Conference established)		
	INVITE(sendonly) 200 OK INVITE(recvonly) ACK ←	← CPG(Isolated)		
	INVITE(sendrecv) ← 200 OK INVITE(sendrecv) → ACK ←	← CPG(Reattached)		
	Conver	rsation ← CPG(Conference disconnected)		
	BYE → 200 OK BYE ←	→ REL ← RLC		

TP504003	SIP reference: RFC 3261 [6]	E	NGN reference: S 283 027 [1], clause 7.4.14			
TSS reference:	SIP-ISUP/SS/CONF/					
SIP selection	NOT PICS 5/10					
criteria:						
ISUP selection						
criteria:						
Test purpose:	No mapping of isolated and reattached					
	Ensure that the SUT on receipt of a CPG m service, the Generic notification indicator w No mapping, no disrupting the SIP proce	th the value				
SIP Parameter	No mapping	, auro.				
values:						
ISUP Parameter	CPG: Generic notification = Conference es	CPG: Generic notification = Conference established				
values:	CPG: Generic notification = isolated					
	CPG: Generic notification = reattached					
	CPG: Generic notification = Conference di					
Comments:	1	JT	ISUP			
	INVITE →	→ ←	IAM ACM			
	100 runging	~	ACIVI			
	Ringing tone 200 OK INVITE	+	ANM			
	ACK →	•	AINIVI			
	-	versation				
	33.11	+	CPG(Conference established)			
		←	CPG(Isolated)			
	Con	← versation	CPG(Reattached)			
	Con	(€	CPG(Conference disconnected)			
	BYE →	→	REL			
	200 OK BYE ←	É				
	<u> </u>	<u>-</u>				

TP504004	SIP reference: RFC 3261 [6]			NGN reference: 7.4.14/[14]
TSS reference:	SIP-ISUP/SS/CONF/			
SIP selection criteria:	PICS [16] 8/2			
ISUP selection criteria:	PICS [16] 5/10			
Test purpose:	No mapping of generic notifications n	o change	e the sess	sion state
	Ensure that the MGCF can receive in added" or "other party isolated" or "otl conference floating" or " other party d side and the call is not disrupted.	her party	reattach	ed" or "other party split" or "
SIP Parameter values:				
ISUP Parameter				
values:				
Comments:	SIP	MGCF		ISUP
	INVITE ->		→	IAM
	180 Ringing ←		(ACM
	200 OK INVITE ← ACK →		+	ANM
			←	CPG(Conference established)
			←	CPG(other party added)
			←	CPG(other party isolated)
			←	CPG(other party reattached)
			←	CPG(other party split)
			←	CPG(other party disconnected)
			←	CPG(Conference floating)
			←	CPG(Conference disconnected)
	BYE ← 200 OK BYE →		← →	REL RLC

TP504005	SIP reference: RFC 3261 [6]		NGN reference: ES 283 027 [1], clause 7.5.6
TSS reference:	SIP-ISUP/SS/CONF/		20 200 027 [1], oladoo 7.0.0
SIP selection	PICS 1/1		
criteria:	1100 1/1		
ISUP selection			
criteria:			
Test purpose:	Conference notification information is mapped	l into "co	nference established"
	Upon the receipt of a conference information of element active is set to "true", the MGCF shall notification "conference established".	l send a	CPG message to the CS side with a
SIP Parameter	NOTIFY 1: Event contains conference; Su	bscriptio	n-State contains active;
values:	expires=xxxx		
	application/conference-info+xm	l:	
	<conference-info></conference-info>		
	entity=conference URI st	tate="full	" version="x"
	<pre><conference-state></conference-state></pre>	oount.	if propert
	<active>true</active>		
	<users></users>	- II pico	Sitt
	<user entity="ISUPx" l<="" th=""><th>JRI state</th><th>="full"</th></user>	JRI state	="full"
	<pre><endpoint entity="</pre"></endpoint></pre>		
	<status>conn</status>		
	<joining-meth< th=""><th>od>diale</th><th>d-out<!-- joining-method--></th></joining-meth<>	od>diale	d-out joining-method
	<media <="" id="1" th=""><th></th><th></th></media>		
	<status>se</status>	endrecv<	
ISUP Parameter			
values:	212		10115
Comments:	SIP MGCF		ISUP
	INVITE -	→	IAM
	180 Ringing ←	(ACM
	200 OK INVITE ←	+	ANM
	ACK →		
	INVITE(SDP focus) →		
	200 OK INVITE		
	ACK →		
	NOTIFY 1 →	→	CPG(Conference established)
	BYE ←	←	REL
	200 OK BYE →	→	RLC
	1		

TP504006	SIP reference: RI	FC 3261 [6]	E	NGN reference: ES 283 027 [1], clause 7.5.6		
TSS reference:	SIP-ISUP/SS/CONF/					
SIP selection criteria:	PICS 1/1					
ISUP selection criteria:						
Test purpose:	Conference notification in	formation is mapp	ed into "oth	er party added"		
	element status of endpoin before and the Contact Ul	nt-status-type is se RI in the element on the MGCF shall se	et to "connec entity is not	t with the <endpoint-type> and the cted" and it was not set to "on-hold" the address of the served nessage to the CS side with a</endpoint-type>		
SIP Parameter	NOTIFY 1: see test case	504005				
values:	application/ <confer entit <con <</con </confer 	NOTIFY 2: Event contains conference ; Subscription-State contains active application/conference-info+xml:				
		user entity=SIPx endpoint entit				
		<status>co</status>	nnected <th>status></th>	status>		
		<joining-me <media id="</th"><th></th><th>d-out<!-- joining-method--></th></media></joining-me 		d-out joining-method		
			sendrecv<	/status>		
ISUP Parameter values:						
Comments:	SIP	MG	CF	ISUP		
	INVITE	→	→	IAM		
	180 Ringing 200 OK INVITE	←	+	ACM ANM		
	ACK	→	•	7.11.11		
	INVITE(SDP focus) 200 OK INVITE ACK	→ ← →				
	NOTIFY 1 200 OK NOTIFY	→ ←	→	CPG(Conference established)		
	NOTIFY 2 200 OK NOTIFY	→ ←	→	CPG(other party added)		
	BYE	←	←	REL		
	200 OK BYE	→	→	RLC		

TP504007	SIP reference: RFC 3261	[6]		ference: 027 [1], clause 7.5.6
TSS reference:	SIP-ISUP/SS/CONF/		1	
SIP selection criteria:	PICS 1/1			
ISUP selection				
criteria:				
Test purpose:	element status of endpoin before and the Contact Ul	erence informates t-status-type is I in the eleme	ation documer set to "on-ho nt entity is the	otated" It with the <endpoint-type> and the ld" and it was set to "connected" address of the served PSTN/ISDN he CS side with a notification</endpoint-type>
SIP Parameter values:	NOTIFY 1: see test case 504005 NOTIFY 2: Event contains conference ; Subscription-State contains active application/conference-info+xml:			
ISUP Parameter		10101	us>sendrecv<	,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,
values:				
Comments:	SIP INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE ACK	→ ← → → +	IGCF → ←	ISUP IAM ACM ANM
	NOTIFY 1 200 OK NOTIFY	→	→	CPG(Conference established)
	NOTIFY 2 200 OK NOTIFY	→ ←	→	CPG(isolated)
	BYE 200 OK BYE	← →	← →	REL RLC

TP504008	SIP reference: RFC 3261	[6]		ference: 027 [1], clause 7.5.6
TSS reference:	SIP-ISUP/SS/CONF/			
SIP selection criteria:	PICS 1/1			
ISUP selection				
criteria:				
Test purpose:	element status of endpoir before and the Contact U	ference informa it-status-type is RI in the eleme ne MGCF shall	tion documen set to "on-hol nt entity is not	t with the <endpoint-type> and the d" and it was set to "connected" the address of the served nessage to the CS side with a</endpoint-type>
SIP Parameter values:	NOTIFY 1: see test case 504005 NOTIFY 2: Event contains conference ; Subscription-State contains active application/conference-info+xml:			
		<media i<="" th=""><th>u= 1 us>sendrecv<</th><th>/otatue></th></media>	u= 1 us>sendrecv<	/otatue>
ISUP Parameter values:		\Stat	uszsenuieuv	/otatus/
Comments:	SIP INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE ACK	M ← ← → ←	GCF → ← ←	ISUP IAM ACM ANM
	NOTIFY 1 200 OK NOTIFY	→	→	CPG(Conference established)
	NOTIFY 2 200 OK NOTIFY	→ ←	→	CPG(other party isolated)
	BYE 200 OK BYE	← →	← →	REL RLC

TP504009	SIP reference: RFC 3261 [6]		-	erence: 027 [1], clause 7.5.6		
TSS reference:	SIP-ISUP/SS/CONF/	SIP-ISUP/SS/CONF/					
SIP selection criteria:	PICS 1/1						
ISUP selection criteria:							
Test purpose:	Conference notification info	Conference notification information is mapped into "reattached"					
	element status of endpoint-s before and the Contact URI participant, the MGCF shall "reattached".	status-type in the elen send a CP	is set to " nent entity	connect is the	t with the <endpoint-type> and the cted" and it was set to "on-hold" address of the served PSTN/ISDN the CS side with a notification</endpoint-type>		
SIP Parameter values:	application/co <conference <use="" entity="<conference"> <use> <use <use=""> <use> <u< th=""><th>s conference-inference-inference-state ser-counts-ser entity=l < endpoint < status < joinin < media ser-counts-ser conference-inference-inference-inference-state ser-counts-ser entity=l < endpoint < status < joinin < status < joinin < media ser entity=l < endpoint < status < joinin < media < media ser entity=l < endpoint < status < joinin < media ser-counts < status < joinin < status < stat</th><th>nfo+xml: e URI state e> 3on-hold g-method a id="1" eatus>sen nfo+xml: e URI state e> 3connec</th><th>e="full" Dunt> if I state= dpoint d dialect drecv if I state= dpoint tted >dialect</th><th>="full" ISUPx URI us> d-out<!-- joining-method--> /status> n-State contains active version="x" f present ="full" ISUPx URI status> d-out<!-- joining-method--></th></u<></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></use></conference>	s conference-inference-inference-state ser-counts-ser entity=l < endpoint < status < joinin < media ser-counts-ser conference-inference-inference-inference-state ser-counts-ser entity=l < endpoint < status < joinin < status < joinin < media ser entity=l < endpoint < status < joinin < media < media ser entity=l < endpoint < status < joinin < media ser-counts < status < joinin < status < stat	nfo+xml: e URI state e> 3on-hold g-method a id="1" eatus>sen nfo+xml: e URI state e> 3connec	e="full" Dunt> if I state= dpoint d dialect drecv if I state= dpoint tted >dialect	="full" ISUPx URI us> d-out joining-method /status> n-State contains active version="x" f present ="full" ISUPx URI status> d-out joining-method		
ISUP Parameter							
values: Comments:	SIP		MGCF		ISUP		
	INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE	→ ← → → ←		→ ← ←	IAM ACM ANM		
	ACK NOTIFY 1 200 OK NOTIFY	→ + +		→	CPG(Conference established)		
	NOTIFY 2 200 OK NOTIFY	→ ←		→	CPG(isolated)		
	NOTIFY 3 200 OK NOTIFY	→ ←		→	CPG(reattached)		
	BYE 200 OK BYE	← →		← →	REL RLC		

TP504010	SIP reference: RFC 3261 [6]]	NGN ref	erence: 027 [1], clause 7.5.6
TSS reference:	SIP-ISUP/SS/CONF/		<u> </u>	
SIP selection criteria:	PICS 1/1			
ISUP selection criteria:				
Test purpose:	Conference notification inforr	nation is mappe	d into "oth	er party reattached"
	element status of endpoint-st before and the Contact URI in	atus-type is set the element er MGCF shall send	to "connec ntity is not	t with the <endpoint-type> and the cted" and it was set to "on-hold" the address of the served nessage to the CS side with a</endpoint-type>
SIP Parameter values:	application/con <conference <conference="" <user<="" <users:="" entity="0" th=""><th>conference; Sufference-info+xme-info> conference URI sence-state> cer-count>3 cer entity=SIPx Usendpoint entity= <status>on-ference; Sufference-info+xme-info> conference URI sence-state> cer-count>3 cer entity=SIPx Usence-state> cer-count>3 cer entity=SIPx User-count>3</status></th><th>nl: state="full" r-count> i RI state=" endpoint nolddialed " sendrecv< ubscription nl: state="full" r-count> i RI state="</th><th>f present Ifull" SIPx URI us> d-out<!-- joining-method--> /status> n-State contains active ' version="x" f present</th></conference>	conference; Sufference-info+xme-info> conference URI sence-state> cer-count>3 cer entity=SIPx Usendpoint entity= <status>on-ference; Sufference-info+xme-info> conference URI sence-state> cer-count>3 cer entity=SIPx Usence-state> cer-count>3 cer entity=SIPx User-count>3</status>	nl: state="full" r-count> i RI state=" endpoint nolddialed " sendrecv< ubscription nl: state="full" r-count> i RI state="	f present Ifull" SIPx URI us> d-out joining-method /status> n-State contains active ' version="x" f present
		endpoint entity: status> con joining-metl media id="1	nected <th></th>	
ISUP Parameter		<status>s</status>	sendrecv<	/status>
values:				
Comments:	SIP INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE ACK	MGCI → ← ← → →	÷ + +	ISUP IAM ACM ANM
	NOTIFY 1	→	→	CPG(Conference established)
	200 OK NOTIFY NOTIFY 2 200 OK NOTIFY	←	→	CPG(other party isolated)
	NOTIFY 3 200 OK NOTIFY	→	→	CPG(other party reattached)
	BYE 200 OK BYE	← →	← →	REL RLC

SIP reference: RFC 3261 [6]		-	erence: 027 [1], clause 7.5.6			
SIP-ISUP/SS/CONF/						
PICS 1/1						
Conference notification informa	ation is mapped	into "oth	er party disconnected"			
element status of endpoint-state method of joining-type is not se	us-type is set to et to "focus-owne	discon er, the M	nected" and the element joining- IGCF shall send a CPG message to			
NOTIFY 2: Event contains of application/conference-entity=consers> <users> <users< users=""> <users> <users< users=""> <users< users=""> <users< users=""> <users< th="" users<=""><th>onference; Sub- prence-info+xml: pinfo> Inference URI stance-state> -count>3conne <joining-metho <media="" <status="" id="1">se onference; Sub- prence-info+xml: pinfo> Inference URI stance-state> -count>3disco- ing-metho <disconnection< th=""><th>ate="full" count> i I state=" endpoint ecteddialect od>dialect obscription I state="full" count> i I state=" endpoint nnected od>dialect od<dialect od="">dialect od>dialect od dialect od dialect od dialect od dialect od dialect od dialect od dialect od dialect</br></br></br></br></br></dialect></th><th>f present full" SIPx URI status> d-out<!-- joining-method--> //status> n-State contains active f version="x" f present full" SIPx URI</th></disconnection<></joining-metho></th></users<></users<></users<></users<></users></users<></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users></users>	onference; Sub- prence-info+xml: pinfo> Inference URI stance-state> -count>3conne <joining-metho <media="" <status="" id="1">se onference; Sub- prence-info+xml: pinfo> Inference URI stance-state> -count>3disco- ing-metho <disconnection< th=""><th>ate="full" count> i I state=" endpoint ecteddialect od>dialect obscription I state="full" count> i I state=" endpoint nnected od>dialect od<dialect od="">dialect od>dialect od dialect od dialect od dialect od dialect od dialect od dialect od dialect od dialect</br></br></br></br></br></dialect></th><th>f present full" SIPx URI status> d-out<!-- joining-method--> //status> n-State contains active f version="x" f present full" SIPx URI</th></disconnection<></joining-metho>	ate="full" count> i I state=" endpoint ecteddialect od>dialect obscription I state="full" count> i I state=" endpoint nnected od>dialect od <dialect od="">dialect od>dialect od dialect od dialect od dialect od dialect od dialect od dialect od dialect od dialect</br></br></br></br></br></dialect>	f present full" SIPx URI status> d-out joining-method //status> n-State contains active f version="x" f present full" SIPx URI			
		ndroov	lototuo			
	<status>se</status>	narecv<	/status>			
SIP INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE ACK NOTIFY 1 200 OK NOTIFY NOTIFY 2 200 OK NOTIFY NOTIFY 3 200 OK NOTIFY	MGCF	→ ← ←	ISUP IAM ACM ANM CPG(Conference established) CPG(other party added) CPG(other party disconnected)			
	SIP-ISUP/SS/CONF/ PICS 1/1 Conference notification information and the receipt of a conference element status of endpoint-station method of joining-type is not set the CS side with a notification." NOTIFY 1: see test case 50400 NOTIFY 2: Event contains capplication/conference-entity=conserved conference-entity=conserved conference-ent	SIP-ISUP/SS/CONF/ PICS 1/1 Conference notification information is mapped Upon the receipt of a conference information of element status of endpoint-status-type is set to method of joining-type is not set to "focus-owner the CS side with a notification "other party dis NOTIFY 1: see test case 504005 NOTIFY 2: Event contains conference; Subsepplication/conference-info+xml:	SIP-ISUP/SS/CONF/ PICS 1/1 Conference notification information is mapped into "oth Upon the receipt of a conference information document element status of endpoint-status-type is set to "disconmethod of joining-type is not set to "focus-owner, the M the CS side with a notification "other party disconnec NOTIFY 1: see test case 504005 NOTIFY 2: Event contains conference; Subscription application/conference-info+xml:			

TP504012	SIP reference: RFC 326	61 [6]		ference: 027 [1], clause 7.5.6
TSS reference:	SIP-ISUP/SS/CONF/			
SIP selection criteria:	NOT PICS 1/1			
ISUP selection criteria:				
Test purpose:	Conference notification	information is n	napped into "oti	her party added"
	information is not mappe	ed to the PSTN		nt the conference notification FY is sent to the ISDN user.
SIP Parameter values:	NOTIFY 1: see test case 504005 NOTIFY 2: Event contains conference ; Subscription-State contains active application/conference-info+xml:			
ISUP Parameter				
values:				
Comments:	INVITE 180 Ringing 200 OK INVITE INVITE(SDP focus) 200 OK INVITE ACK	→ ← ← → ←	MGCF → ←	ISUP IAM ACM ANM
	NOTIFY 1 200 OK NOTIFY	→	→	CPG(Conference established)
	NOTIFY 2 200 OK NOTIFY	→ ←	→	CPG(other party added)
	BYE 200 OK BYE	← →	← →	REL RLC

TP504013	SIP reference: RFC 32	261 [6]	I-		erence: 027 [1], clause 7.5.6
TSS reference:	SIP-ISUP/SS/CONF/		•		
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	The referring of MGCF Ensure that a REFER I rejected with 403 Forbi	request receiv	ed by the I	MGCF is	s not successful. The request is
SIP Parameter values:	RÉFER: Request URI Refer-To cont	contained the calcium the URI of contains SIP or t	onference I ISUPx, met	URI hod=invi	
ISUP Parameter values:					
Comments:	SIP		MGCF		ISUP
	INVITE	→		→	IAM
	180 Ringing	←		←	ACM
	200 OK INVITE	←		←	ANM
	ACK	→			
	REFER 403 Forbidden	→			

TP504014	SIP reference: RFC 3261 [6]	1 N	NGN reference:			
	-	Ē	S 283 027 [1], clause 7.5.6			
TSS reference:	SIP-ISUP/SS/CONF/					
SIP selection criteria:						
ISUP selection criteria:						
Test purpose:	Ensure that a REFER reques	The referring of MGCF is not possible call is not established Ensure that a REFER request received by the MGCF is not successful. The request is rejected with 403 Forbidden. The CS -site is not affected.				
SIP Parameter values:		ned the conference L e URI of ISUPx , meth s SIP or tel URI of SII	nod=invite			
ISUP Parameter values:						
Comments:	SIP REFER 403 Forbidden	MGCF → ←	ISUP			

6.3.1.5 Three Party service (3PTY)

TP505001	SIP reference: RFC 3261 [6	6]	-	NGN reference:
TSS reference:	CID ICUD/CC/2DTV/		E.	S 283 027 [1], clause 7.4.15
SIP selection	SIP-ISUP/SS/3PTY/			
criteria:	PICS 8/2			
ISUP selection	PICS 5/5 AND PICS 5/18			
criteria:				
Test purpose:	Notification procedure supported			
	Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the 3PTY supplementary service. • If the media stream is either in state "sendonly" or "inactive" then: INVITE with the			
	attribute line a_LINE_VA, o		attribute l	ine, else: no mapping.
SIP Parameter	SDP: a= a_LINE_VA (see table 38	3)		
values:				
ISUP Parameter	CPG: notification = remote hold			
values:	CPG: Generic notification = GEN_I		JE	
Comments:	SIP	SUT	_	ISUP
	INVITE →			IAM
	180 Ringing ←		+	ACM
	Ringing tone			
	200 OK INVITE ←		+	ANM
	ACK →			
		Conver		
	INVITE(sendonly)		←	CPG(hold)
	200 OK INVITE(recvonly) →			
	ACK +			
	INVITE(sendrecv) ←		←	CPG(Conference established)
	200 OK INVITE(sendrecv) →			
	ACK ←			
		Conver	sation	
			←	CPG(Conference disconnected)
		Conver	sation	
	BYE →		→	REL
	200 OK BYE ←	-1		RLC

Table 38

	Values for test purpose TP505001				
	←INVITE/UPDATE	← CPG			
	SDP attribute line	Generic notification			
	a_LINE_VA	GEN_NOT_VALUE			
VA_01	sendonly or inactive	Conference established			
VA_02	sendrecv or recvonly or omitted	Conference disconnected			

TP505002	SIP reference: RFC 3261 [6]		NGN reference: 3 283 027 [1], clause 7.4.15 TRec Q.734.2 [35], clause 2.7	
TSS reference:	SIP-ISUP/SS/3PTY/			
SIP selection				
criteria:				
ISUP selection criteria:	NOT PICS 5/18			
Test purpose:	Notification procedure not supported			
	Ensure that the SUT on receipt of a CPG message due to the 3PTY supplementary service, the Generic notification indicator with the value. No mapping, no disrupting the SIP procedure.			
SIP Parameter	No mapping			
values:				
ISUP Parameter	CPG: Generic notification = Conference esta			
values: Comments:	CPG: Generic notification = Conference disc		ISUP	
Comments.	INVITE +		IAM	
		-	ACM	
	180 Ringing ← Ringing tone	~	ACIVI	
	Kinging tone	←	CPG(hold)	
			CPG(Conference established)	
	Conve	rsation		
		←	CPG(Conference disconnected)	
		rsation	DE!	
	BYE ->	→	REL	
	200 OK BYE ←		RLC	

TP504003	SIP reference: RFC 3	3261 [6]	E	NGN reference: S 283 027 [1], clause 7.4.15	
TSS reference:	SIP-ISUP/SS/3PTY				
SIP selection criteria:	PICS 1/1				
ISUP selection criteria:					
Test purpose:	Conference notification information is mapped into "conference established" Upon the receipt of a conference information document with the <conference-state-type> element active is set to "true", the MGCF shall send a CPG message to the CS side with a notification "conference established".</conference-state-type>				
SIP Parameter values:	NOTIFY 1: Event contains conference; Subscription-State contains active; expires=xxxx application/conference-info+xml: <conference-info> entity=conference URI state="full" version="x" <conference-state> <user-count>2</user-count> if present <active>true</active> if present <users> <user <endpoint="" <status="" entity="endpoint" isupx="" state="full" uri="">connected <joining-method>dialed-out</joining-method> <media <status="" id="1">sendrecv</media></user></users></conference-state></conference-info>				
ISUP Parameter					
values:			_		
Comments:	INVITE 180 Ringing 200 OK INVITE ACK	MGC → ← ← →	F → ← ←	ISUP IAM ACM ANM	
	INVITE(sendonly) 200 OK INVITE(recvonly) ACK INVITE(SDP focus) 200 OK INVITE ACK	→ → → →	→	CPG(hold)	
	NOTIFY 1	→	→	CPG(Conference established)	
	BYE 200 OK BYE	← →	←	REL RLC	

6.3.1.6 Connected line identification (COL)

TP506001			=	SUP reference:
			[15] clause 7.4.2
TSS reference:	SIP-ISUP/SS/COL/			
SIP selection criteria:	NOT PICS 5/22			
ISUP selection criteria:				
Test purpose:	Mapping of connected r	number not suppoi	ted	
	Ensure that the SUT, if a signalling procedure. The			ANM, does not disrupt the SIP to any SIP message.
SIP Parameter				
values:				
ISUP Parameter	ANM: Connected numb	er Parameter		
values:				
Comments:	SIP		MGCF	ISUP
	INVITE	→	→	IAM
	180 Ringing	←	(ACM
	200 OK INVITE	←	+	ANM
	ACK	→		
		Co	nversation	
	BYE	→	→	REL
	200 OK BYE	É	+	RLC

TP506002			JP reference: uses 7.4.2 and 7.5.2	
TSS reference:	SIP-ISUP/SS/COL/			
SIP selection	PICS 5/22 AND PICS 13/1			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Connected number national, presentation allo	wed, no additional	connected number received	
	Ensure that the SUT, on receipt of an ANM m	essage with a		
	Connected number parameter coded			
	Address presentation restricted parameter = p		ed	
	Nature of address indicator = national number			
	Numbering plan indicator = ISDN/Telephony r	iumbering plan		
	Screening indicator = Network provided			
	Address signals in the format: NDC+SN			
	and without the Generic number parameter,			
	sends a 200 OK INVITE to the UAC with a			
	P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and Add CC (of the country where the MGCF is located) to Connected PN address signals to construct E.164 number in URI global number format.			
SIP Parameter	200 OK INVITE: P-Asserted-Identity header fi	eld Tel URL contai	ning an URI in the format	
values:	"+"CC+NDC+SN			
ISUP Parameter	ANM;			
values:	Connected number parameter			
	Address presentation restricted parameter = '(10.R		
	Nature of address indicator = '0000011'B Numbering plan indicator = '001'B			
	Screening indicator = Network provided			
	Address signals = derrived from the P-Asserte	d-Identity		
	Generic number parameter not present	a raditity		
Comments:	•	GCF	ISUP	
	INVITE -	→	IAM	
	180 Ringing ←	É	ACM	
	200 OK INVITE	÷	ANM	
	ACK →	-		
		rsation		
	BYE →	→	REL	
	200 OK BYE	É	RLC	
	200 OK BIL		ILLO	

TP506003				GN reference:], clauses 7.4.2 and 7.5.2
TSS reference:	SIP-ISUP/SS/COL/		20 200 027 [1	j, oladooo 7.4.2 and 7.0.2
SIP selection	PICS 5/22 AND PICS 13/1			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Connected number international, presentation allowed, no additional connected number received			
	Ensure that the SUT, on receipt of	an ANM me	essage with a	
	Connected number parameter co	oded		
	Address presentation restricted pa		resentation allowe	ed
	Nature of address indicator = interi			
	Numbering plan indicator = ISDN/7		umbering plan	
	Screening indicator = Network prov			
	Address signals in the format: CC+	NDC+SN		
	and without the Generic number p	oarameter,		
	sends a 200 OK INVITE to the UAC with a			
	P-Asserted-Identity header field NDC+ SN as received in the co	nnected nu	mber in the ANM.	•
SIP Parameter values:	200 OK INVITE: P-Asserted-Identification "+"CC+NDC+SN	ty header fie	eld Tel URL conta	ining an URI in the format
ISUP Parameter	ANM:			
values:	Connected number parameter			
values.	Address presentation restricted parameter = '00'B			
	Nature of address indicator = 0000		,0 B	
	Numbering plan indicator = '001'B	.002		
	Screening indicator = Network prov	/ided		
	Address signals = PIXIT			
	Generic number parameter not p	resent		
Comments:	SIP	M	GCF	ISUP
	INVITE -	→	→	IAM
		-	←	ACM
		-	←	ANM
	ACK -	→		
		Conve	rsation	
	BYE -	→	→	REL
	200 OK BYE	-	←	RLC
			<u>,</u>	

TP506004			JP reference:	
TCC reference:		[14] claus	ses 7.4.2.2 and 7.5.2	
TSS reference: SIP selection	SIP-ISUP/SS/COL/ PICS 5/22 AND NOT PICS 13/1			
criteria:	PICS 5/22 AND NOT PICS 13/1			
ISUP selection criteria:				
Test purpose:	Connected number national, presentation allowed, additional connected number received			
	Ensure that the SUT, on receipt of an ANM message with a			
	Connected number parameter coded Address presentation restricted parameter = presentation allowed Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals in the format: NDC+SN			
	Generic number parameter, Number Qualifier Indicator "Additional connected number" Address presentation restricted parameter = presentation allowed Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT NDC+SN			
	sends a 200 OK INVITE to the UAC with a			
	P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and Add CC (of the country where the MGCF is located) to Connected PN address signals to construct E.164 number in URI. Prefix number with "+".			
010.0	The additional connected number is not interworked			
SIP Parameter	200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format			
values:	"+"CC+NDC+SN			
ISUP Parameter	ANM;			
values:	Connected number parameter Address presentation restricted parameter = Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = Network provided Address signals = PIXIT Generic number parameter Number Qualifier Indicator "00000101"B Address presentation restricted parameter = Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT	00'B	IOUD	
Comments:		IGCF	ISUP	
	INVITE -	→	IAM	
	180 Ringing ←	←	ACM	
	200 OK INVITE ←	←	ANM	
	ACK →			
	Conv	ersation		
	BYE →	→	REL	
	200 OK BYE ←	+	RLC	

TP506005		ISUP reference:		
		[14] clauses 7.4.2.2 and 7.5.2		
TSS reference:	SIP-ISUP/SS/COL/			
SIP selection	PICS 5/22 AND NOT PICS 13/1			
criteria:				
ISUP selection criteria:				
Test purpose:	Connected number international, presentation allowed, additional connected number			
rest purpose.	received			
	Ensure that the SUT, on receipt of an ANM m	nessage with a		
	Connected number parameter coded			
	Address presentation restricted parameter =			
	Nature of address indicator = international nu			
	Numbering plan indicator = ISDN/Telephony	numbering plan		
	Screening indicator = Network provided	CLON		
	Address signals in the format: PIXIT CC+ND0	2+3N		
	Generic number parameter,			
	Number Qualifier Indicator "Additional connection	cted number"		
	Address presentation restricted parameter =	presentation allowed		
	Nature of address indicator = international nu			
	Numbering plan indicator = ISDN/Telephony			
	Screening indicator = user provided, not verif	ied		
	Address signals = CC+NDC+SN			
	sends a 200 OK INVITE to the UAC with a			
	P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+			
	NDC+ SN has been received and the			
	The additional connected number is not interworked			
SIP Parameter	200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format			
values:	"+"CC+NDC+SN			
ISUP Parameter	ANM;			
values:	Connected number parameter			
		Address presentation restricted parameter = '00'B		
	Nature of address indicator = "0000100'B			
	Numbering plan indicator = '001'B Screening indicator = Network provided			
	Address signals = PIXIT			
	Generic number parameter			
	Number Qualifier Indicator "00000101"B			
	Address presentation restricted parameter =	'00'B		
	Nature of address indicator = '0000100'B			
	Numbering plan indicator = '001'B			
	Screening indicator = '00'B			
Commonsta	Address signals = PIXIT	ACCE ICUD		
Comments:		MGCF ISUP		
	INVITE ->	→ IAM		
	180 Ringing ← 200 OK INVITE ←	← ACM ← ANM		
	200 OK INVITE ← ACK →	AINIVI		
		ersation		
	BYE -	ersation → REL		
	200 OK BYE	← RLC		
	200 ON DIL	· INLO		

TP506006		ISUP reference:			
		[14] clauses 7.4.2 and 7.5.2			
TSS reference:	SIP-ISUP/SS/COL/				
SIP selection	PICS 5/22 AND NOT PICS 13/1				
criteria:					
ISUP selection criteria:					
Test purpose:	Connected number national, presentation restricted, no additional connected number received				
	Ensure that the SUT, on receipt of an ANM message with a				
	Connected number parameter coded				
	Address presentation restricted parameter = p	resentation restricted			
	Nature of address indicator = national number				
	Numbering plan indicator = ISDN/Telephony n	numbering plan			
	Screening indicator = Network provided				
	Address signals in the format: PIXIT NDC+SN				
	and without the Generic number parameter,				
	sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format "+" CO NDC+ SN has been received and Add CC (of the country where the MGCF is located Connected PN address signals to construct E.164 number in URI. Prefix number with "+". a Privacy header is inserted with the value "id" or the value "id" is added to a existence.				
OID D	Privacy header				
SIP Parameter	200 OK INVITE: P-Asserted-Identity header fie	eld Tel URL containing an URI in the format			
values:	"+"CC+NDC+SN				
ISUP Parameter values:	ANM;				
values.	Connected number parameter Address presentation restricted parameter = '0	םיות			
	Nature of address indicator = '0000011'B	71 0			
	Numbering plan indicator = '001'B				
	Screening indicator = Network provided				
	Address signals = PIXIT				
	Generic number parameter not present				
Comments:		GCF ISUP			
	INVITE →	→ IAM			
	180 Ringing ←	← ACM			
	200 OK INVITE ←	← ANM			
	ACK →				
		rsation			
	BYE ->	→ REL			
	200 OK BYE	← RLC			

TP506007		ISUP reference:			
		[14] clauses 7.4.2 and 7.5.2			
TSS reference:	SIP-ISUP/SS/COL/				
SIP selection	PICS 5/22 AND NOT PICS 13/1				
criteria:					
ISUP selection					
criteria:					
Test purpose:	Connected number international, presentation restricted, no additional connected number received				
	Ensure that the SUT, on receipt of an ANM message with a				
	Connected number parameter coded				
	Address presentation restricted parameter = p				
	Nature of address indicator = international nur				
	Numbering plan indicator = ISDN/Telephony r	numbering plan			
	Screening indicator = Network provided	N. ONI			
	Address signals in the format: PIXIT CC+NDC	+5N			
	and without the Generic number parameter ,				
	sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the				
		"id" or the value "id" is added to a existence			
	Privacy header				
SIP Parameter	200 OK INVITE: P-Asserted-Identity header fi	eld Tel URL containing an URI in the format			
values:	"+"CC+NDC+SN	·			
ISUP Parameter	ANM;				
values:	Connected number parameter				
	Address presentation restricted parameter = '0)1'B			
	Nature of address indicator = 0000100'B				
	Numbering plan indicator = '001'B				
	Screening indicator = Network provided				
	Address signals = PIXIT Generic number parameter not present				
Comments:		GCF ISUP			
Comments:	INVITE →	GCF ISUP → IAM			
	·····	→ IAIVI ← ACM			
	180 Ringing ← 200 OK INVITE ←	← ANM			
	ACK →	AINIVI			
	7.5.1	ersation			
	BYE →	→ REL			
	200 OK BYE	← RLC			
	ZUU UN DIE	T NLU			

TP506008		IS	SUP reference:		
		[14] cla	uses 7.4.2 and 7.5.2		
TSS reference:	SIP-ISUP/SS/COL/				
SIP selection criteria:	PICS 5/22 AND NOT PICS 13/1				
ISUP selection criteria:					
Test purpose:	Connected number national, presentation restricted, additional connected number received				
	Ensure that the SUT, on receipt of an ANM	message with a			
	Connected number parameter coded Address presentation restricted parameter :	- procentation roots	atad		
	Nature of address indicator = national number		cieu		
	Numbering plan indicator = ISDN/Telephon	y numbering plan			
	Screening indicator = Network provided				
	Address signals in the format: PIXIT NDC+	SN			
	Generic number parameter, Number Qualifier Indicator "Additional conn	ected number"			
	Address presentation restricted parameter =		cted		
	Nature of address indicator = national number	er			
	Numbering plan indicator = ISDN/Telephon				
	Screening indicator = user provided, not verified Address signals = NDC+SN				
	sends a 200 OK INVITE to the UAC with a				
	P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and Add CC (of the country where the MGCF is located) to Connected PN address signals to construct E.164 number in URI. Prefix number with "+".				
	a Privacy header is inserted with the val		"id" is added to a existence		
SIP Parameter	The additional connected number is not 200 OK INVITE: P-Asserted-Identity header		nining on LIDI in the format		
values:	"+"CC+NDC+SN	neid Tei URL conta	aining an ORI in the format		
ISUP Parameter	ANM;				
values:	Connected number parameter				
	Address presentation restricted parameter = '01'B				
	Nature of address indicator = '0000011'B Numbering plan indicator = '001'B				
	Screening indicator = Network provided				
	Address signals = PIXIT				
	Generic number parameter				
	Number Qualifier Indicator "00000101"B				
	Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B				
	Numbering plan indicator = '001'B				
	Screening indicator = '00'B				
Commonts	Address signals = PIXIT SIP	MGCF	ICIID		
Comments:	INVITE -	MGCF →	ISUP IAM		
	180 Ringing	-	ACM		
	200 OK INVITE	÷	ANM		
	ACK →				
		versation			
	BYE -	→	REL		
	200 OK BYE ←		RLC		

TP506009				SUP reference: auses 7.4.2 and 7.5.2	
TSS reference:	SIP-ISUP/SS/COL/				
SIP selection	PICS 5/22 AND NOT PICS 13/	1			
criteria:					
SUP selection					
criteria:	Connected number international, presentation restricted, additional connected num				
Test purpose:	received	ai, presentation	restricted, addit	ionai connected number	
	Ensure that the SUT, on receip	ot of an ANM mo	essage with a		
	Connected number parameter Address presentation restricted		resentation restr	icted	
	Nature of address indicator = ir			icieu	
	Numbering plan indicator = ISE				
	Screening indicator = Network		31		
	Address signals in the format: I		+SN		
	Generic number parameter,	ditional connec	tod numbor"		
	Number Qualifier Indicator "Ad			icted	
	Address presentation restricted parameter = presentation restricted Nature of address indicator = international number				
	Numbering plan indicator = ISDN/Telephony numbering plan				
	Screening indicator = user provided, not verified				
	Address signals = CC+NDC+SN				
	sends a 200 OK INVITE to the UAC with a				
	NDC+ SN has been receive	ed and the d with the value	"id" or the value	dentity in the format "+" CC+ "id" is added to a existence	
SIP Parameter	200 OK INVITE: P-Asserted-Id	entity header fi	eld Tel URL cont	aining an URI in the format	
/alues:	"+"CC+NDC+SN				
SUP Parameter	ANM;				
/alues:	Connected number paramete		MID		
	Address presentation restricted		שוט		
	Nature of address indicator = "0000100'B Numbering plan indicator = '001'B				
	Screening indicator = Network provided				
	Address signals = PIXIT				
	Generic number parameter				
	Number Qualifier Indicator "00000101"B				
	Address presentation restricted parameter = '001B				
	Nature of address indicator = '0000100'B Numbering plan indicator = '001'B				
	Screening plan Indicator = '00' B				
	Address signals = PIXIT				
Comments:	SIP	M	GCF	ISUP	
	INVITE	→	→	IAM	
	180 Ringing	←	←	ACM	
	200 OK INVITE	←	←	ANM	
	ACK	→			
	5.45		rsation	55	
	BYE	→	→	REL	
	200 OK BYE	<u> </u>		RLC	

	Values for test purposes TP102006-TP102009
VA_01	ISUP_SI = user provided verified and passed, '01'B
VA_02	ISUP_SI = network provided, '11'B

TP506010			ISL	JP reference:
		[14	II clau	ses 7.4.2 and 7.5.2
TSS reference:	SIP-ISUP/SS/COL/	<u> </u>	•	
SIP selection criteria:	PICS 5/22 AND PICS 13/1			
ISUP selection criteria:				
Test purpose:	IAM connected line request indication is	sent		
	Ensure that a optional forward call indicator value Connected line identity request indicator is set to "requested" is contained in the sent IAM if an INVITE request is received containing a Supported header equal to "from-change".			
SIP Parameter	INVITE: Supported: "from-change"			
values:				
ISUP Parameter	IAM: oFCi Connected line identity reque	st indicator is set t	to "req	uested"
values:				
Comments:	SIP	MGCF		ISUP
	INVITE →		→	IAM
	180 Ringing ←		←	ACM
	200 OK INVITE ←		←	ANM
	ACK →			
		Conversation		
	BYE →		→	REL
	200 OK BYE ←		←	RLC

TP506011		ISUP reference:			
		[14] clauses 7.4.2 and 7.5.2			
TSS reference:	SIP-ISUP/SS/COL/				
SIP selection	PICS 5/22 AND PICS 13/1				
criteria:					
ISUP selection					
criteria:					
Test purpose:	Additional connected number national, presentation allowed in ANM is received				
	Ensure that if a ANM is received and a Additional connected number "national number",				
	"prestentation allowed" is included then a 200	OK INVITE is sent and the Supported header			
	contains the "from-change" tag.				
SIP Parameter	200 OK INVITE: Supported: "from-change"				
values:		from the Connected number			
	UPDATE: From header contains the Generic				
ISUP Parameter	IAM: oFCi Connected line identity request ind	cator is set to "requested"			
values:	ANM:				
	Additional connected number				
	Number Qualifier Indicator "00000101"B				
	Address presentation restricted parameter = p				
	Nature of address indicator = national number				
	Numbering plan indicator = ISDN/Telephony r				
	Screening indicator = user provided, not verifi	ea			
	Address signals = PIXIT Connected number parameter				
	Address presentation restricted parameter = p	vrocentation allowed			
	Nature of address indicator = national number				
	Numbering plan indicator = ISDN/Telephony r				
	Screening indicator = Network provided	idilibering plan			
	Address signals = PIXIT				
Comments:		GCF ISUP			
	INVITE →	→ IAM			
	180 Ringing	← ACM			
	200 OK INVITE	← ANM			
	ACK →	7,1110			
	UPDATE ←				
	200 OK UPDATE →				
	Conve	ersation			
	BYE →	→ REL			
	200 OK BYE ←	← RLC			

TSS reference: SIP-ISUP/SS/COL/ SIP selection criteria: ISUP selection criteria: ISUP selection criteria: Test purpose: Additional connected number international, presentation allowed in ANM is received Ensure that if a ANM is received and a Additional connected number "international number" "prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag. SIP Parameter values: P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter values: AMM: Additional connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE	TP506012		ISUP reference:		
SIP selection criteria: ISUP selection criteria: Test purpose: Additional connected number international, presentation allowed in ANM is received Ensure that if a ANM is received and a Additional connected number "international number" prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" 1200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter values: IAM: oFCI Connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address presentation restricted parameter = presentation allowed Nature of address indicator = ISDN/Telephony numbering plan Screening indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Connected number parameter Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE ACM ACM ACM ACM ACM ACM ACM AC			[14] clauses 7.4.2 and 7.5.2		
criteria: ISUP selection criteria: Test purpose: Additional connected number international, presentation allowed in ANM is received Ensure that if a ANM is received and a Additional connected number "international number" "prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag. SIP Parameter values: 200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter values: IAM: oFCi Connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ACM ACM					
Sup selection criteria:	SIP selection	PICS 5/22 AND PICS 13/1			
Test purpose: Additional connected number international, presentation allowed in ANM is received Ensure that if a ANM is received and a Additional connected number "international number" "prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag. SIP Parameter values: 200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter values: IAM: oFCi Connected line identity request indicator is set to "requested" AMM: Additional connected number Number Qualifier Indicator "000001011"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Connected number parameter Numbering plan indicator = Network provided Address signals = PIXIT SIP MGCF ISUP INVITE					
Test purpose: Additional connected number international, presentation allowed in ANM is received Ensure that if a ANM is received and a Additional connected number "international number" "prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag. SIP Parameter values: 200 OK INVITE: Supported: "from-change"					
Ensure that if a ANM is received and a Additional connected number "international number" "prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag. SIP Parameter values: 200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter Values: IAM: oFCi Connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = USDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE	*********				
"prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag. SIP Parameter values: 200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter values: IAM: oFCi Connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE 180 Ringing CMCK ANM ACK ANM ACK ANM	Test purpose:	Additional connected number international, presentation allowed in ANM is received			
SIP Parameter values: 200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter values: IAM: oFCi Connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP		Ensure that if a ANM is received and a Additional connected number "international number"			
SIP Parameter values: 200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter values: IAM: oFCi Connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE			OK INVITE is sent and the Supported header		
P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN ISUP Parameter values: IAM: oFCi Connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE → IAM 180 Ringing ← ACM ACK → ANM					
UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN					
ISUP Parameter values: IAM: oFCi Connected line identity request indicator is set to "requested" ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE IAM 180 Ringing ACK ACM ACK ANM	values:				
ANM: Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE ACM 180 Ringing ACK ANM ACK					
Additional connected number Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE IAM 180 Ringing ACK ACM 200 OK INVITE ANM ACK		, ,	icator is set to "requested"		
Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE ACM 200 OK INVITE ACM ACK ANM	values:	1 11 11 11 11 11 11 11 11 11 11 11 11 1			
Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE ACM 200 OK INVITE ACM ACK ANM					
Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE IAM 180 Ringing ACM 200 OK INVITE ANM ACK ACK ANM					
Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE IAM 180 Ringing ACM 200 OK INVITE ANM ACK ANM					
Screening indicator = user provided, not verified Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE IAM 180 Ringing ACM 200 OK INVITE ANM ACK ACM					
Address signals = PIXIT Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE INVITE IAM 180 Ringing ACM 200 OK INVITE ANM ACK ANM					
Connected number parameter Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE INVITE IAM 180 Ringing ACM 200 OK INVITE ANM ACK ACK			ea		
Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE INVITE IAM 180 Ringing ACM 200 OK INVITE ANM ACK					
Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP MGCF ISUP INVITE HAM 180 Ringing ACM 200 OK INVITE ANM ACK MGCF ACM ANM			proportation allowed		
Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT Comments: SIP INVITE → IAM 180 Ringing ← COM ACK MGCF ISUP INVITE → ACM ACM ACM ANM ACK					
Screening indicator = Network provided Address signals = PIXIT Comments: SIP INVITE + IAM 180 Ringing COMMENTE ACM ACK MGCF ISUP IAM ACK MGCF ACM ACM ACM ACM ANM ACK					
Address signals = PIXIT Comments: SIP MGCF ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ANM ACK →			numbering plan		
SIP MGCF ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ANM ACK → ANM					
INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ANM ACK →	Comments:		IGCE ISUP		
180 Ringing ← ← ACM 200 OK INVITE ← ANM ACK →	Comments.				
200 OK INVITE ← ANM ACK →					
ACK →					
UPDATE ←			ANIVI		
		UPDATE ←			
200 OK UPDATE →		200 OK UPDATE →			
Conversation			ersation		
BYE → REL		BYE →	→ REL		
200 OK BYE					

TP506013		ISUP reference:		
		[14] clauses 7.4.2 and 7.5.2		
TSS reference:	SIP-ISUP/SS/COL/			
SIP selection	PICS 5/22 AND PICS 13/1			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Additional connected number national, presentation restricted in ANM is received			
	Ensure that if a ANM is received and a Addition			
	"prestentation restricted" is included then a 200 OK INVITE is sent and the Supported			
OID Danamatan	header contains the "from-change" tag.			
SIP Parameter	200 OK INVITE: Supported: "from-change"	from the Connected number in the format		
values:	"+"CC+NDC+SN Privacy: id			
	UPDATE: From header contains the Generic			
	Privacy: header			
ISUP Parameter	IAM: oFCi Connected line identity request ind	icator is set to "requested"		
values:	ANM:	·		
	Additional connected number			
	Number Qualifier Indicator "00000101"B			
	Address presentation restricted parameter = p			
	Nature of address indicator = national number			
	Numbering plan indicator = ISDN/Telephony r			
	Screening indicator = user provided, not verifi	ed		
	Address signals = PIXIT			
	Connected number parameter	area antation reatricts d		
	Address presentation restricted parameter = p Nature of address indicator = national number			
	Numbering plan indicator = ISDN/Telephony r			
	Screening indicator = Network provided	idilibering plan		
	Address signals = PIXIT			
Comments:	Ÿ	GCF ISUP		
	INVITE ->	→ IAM		
	180 Ringing ←	← ACM		
	200 OK INVITE	← ANM		
	ACK →			
	UPDATE ←			
	200 OK UPDATE →			
		ersation		
	BYE →	→ REL		
	200 OK BYE ←	€ RLC		

TP506014		ISUP reference:		
		[14] clauses 7.4.2 and 7.5.2		
TSS reference:	SIP-ISUP/SS/COL/			
SIP selection criteria:	PICS 5/22 AND PICS 13/1			
ISUP selection criteria:				
Test purpose:	Additional connected number international, presentation restricted in ANM is received			
		onal connected number "international number" 0 OK INVITE is sent and the Supported header		
SIP Parameter	200 OK INVITE: Supported: "from-change"			
values:		from the Connected number in the format		
	UPDATE: From header contains the Generic Privacy: header			
ISUP Parameter	IAM: oFCi Connected line identity request ind	icator is set to "requested"		
values:	ANM: Additional connected number			
	Number Qualifier Indicator "00000101"B			
	Address presentation restricted parameter = p			
	Nature of address indicator = international nul			
	Numbering plan indicator = ISDN/Telephony r			
	Screening indicator = user provided, not verifi Address signals = PIXIT	ea		
	Connected number parameter			
	Address presentation restricted parameter = p	presentation restricted		
	Nature of address indicator = international nul	mher		
	Numbering plan indicator = ISDN/Telephony r			
	Screening indicator = Network provided	idinooning plan		
	Address signals = PIXIT			
Comments:		GCF ISUP		
	INVITE →	→ IAM		
	180 Ringing ←	← ACM		
	200 OK INVITE ←	← ANM		
	ACK →			
	UPDATE ←			
	200 OK UPDATE →			
	Conve	ersation		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

6.3.1.7 Malicious call identification MCID

TP507001	SIP reference	e: RFC 3261 [6]		ISUP reference:
			ES 2	83 027 [1], clause 7.4.4
TSS reference:	SIP-ISUP/SS/MCID/			
SIP selection	PICS 9/1			
criteria:				
ISUP selection				
criteria:				
Test purpose:	No interworking MG	CF sends IRS		
				essage. The MCID response
	indicator is set to "Me	CID not included". The SI	P signalling p	rocedure is not disrupted.
SIP Parameter	No influence			
values:				
ISUP Parameter	IDR: MCID reques	ted		
values:	IRS: MCID not incl	uded		
Comments:	SIP	SU	JT	ISUP
	INVITE	→	→	IAM
			←	IDR
			→	IRS
	180 Ringing	←	←	ACM
		Ringing tone		
	200 OK INVITE	←	←	ANM
	ACK	→		
		Conve	rsation	
	BYE	+	+	REL
	200 OK BYE	→	→	RLC

TP507002	SIP reference	ce: RFC 3261 [6]		ISUP re	eference:
			ES 2	83 027 [1], clause 7.4.4
TSS reference:	SIP-ISUP/SS/MCID/	r.			
SIP selection	NOT PICS 9/1				
criteria:					
ISUP selection					
criteria:					
Test purpose:	No interworking time	out T39			
		if an IDR is received, no	IDR is sent. T	he SIP	signalling procedure is
	not disrupted.				
SIP Parameter	No influence				
values:					
ISUP Parameter	IDR: MCID reques	ted			
values:					
Comments:	SIP	SU		ISUP	
	INVITE	→	→	IAM	
			←	IDR	
					T39 timeout
	180 Ringing	←	←	ACM	
		Ringing tone			
	200 OK INVITE	←	←	ANM	
	ACK	→			
		Conver	sation		
	BYE	←	←	REL	
	200 OK BYE	→	→	RLC	

6.3.1.8 Sub-addressing (SUB)

TP508001	SUB Reference: ES 283 027 [1], clause 7.4.5	Selection criteria: PICS 5/8		
TSS reference:	SIP-ISUP/SS/SUB/			
Preconditions:				
Test purpose:	The isub parameter of the P-Asserted-Identity header in an INVITE is mapped in the calling party subaddress in the IAM			
	Ensure that the isub parameter in the P-Asserted-Identity header of the received INVITE is interworked in the Calling party subaddress contained in an ATP parameter in the sent IAM. The Type of Subbaddress is set set to "0 0 0" "NSAP (ITU-T Recommendation X.213 [29]			
SIP Parameter	and ISO/IEC 8348 [30] Add.2)"			
values:	P-Asserted-Identity: sip: user part; isub= <s< td=""><td>subaddress>@hostportion</td></s<>	subaddress>@hostportion		
ISUP Parameter values:	IAM: ATP(Calling party subaddress)	·		
Comments:	SIP	MGCF ISUP		
	INVITE -	→ IAM		
	100 Trying ←			
	180 Ringing ←	← ACM		
	200 OK INVITE ←	← ANM		
	ACK →			
	Commu	nication		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

TP508002	SUB Reference:	Selection criteria:		
17300002	ES 283 027 [1], clause 7.4.5	PICS 5/8		
TSS reference:	SIP-ISUP/SS/SUB/			
Preconditions:				
Test purpose:	The isub parameter of the Request URI in an INVITE is mapped in the called party subaddress in the IAM			
	Ensure that the isub parameter in the Request URI of the received INVITE is interworked is the Called party subaddress contained in an ATP parameter in the sent IAM. The Type of Subbaddress is set set t "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"			
SIP Parameter values:	INVITE: sip: user part; isub= <subaddress>@h</subaddress>	INVITE: sip: user part; isub= <subaddress>@hostportion</subaddress>		
ISUP Parameter	IAM: ATP(Called party subaddress)			
values:	IAW. ATP(Called party subaddress)			
Comments:	SIP	MGCF ISUP		
	INVITE →	→ IAM		
	100 Trying ←			
	180 Ringing ←	← ACM		
	200 OK INVITE ←	← ANM		
	ACK →			
	Commun	nication		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

TP508003	SUB Reference: ES 283 027 [1], clause 7.4.5	Selection criteria: PICS 5/8	
TSS reference:	SIP-ISUP/SS/SUB/		
Preconditions:			
Test purpose:	The connected subaddress in the ANM is mapped in the isub parameter of the P-Asserted- Identity header in the 200 OK INVITE		
	Ensure that the isub parameter in the P-Asserted-Identity header of the received 200 OK INVITE is interworked in the connected subaddress contained in an ATP parameter in the sent ANM. The Type of Subbaddress is set set to "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"		
SIP Parameter	200 OK INVITE:		
values:	P-Asserted-Identity: sip: user part; isub=<	subaddress>@hostportion	
ISUP Parameter	IAM: oFCi: connected line request	- Caracana Constitution	
values:	ANM: ATP(Connected subaddress)		
Comments:	SIP	MGCF ISUP	
	INVITE +	→ IAM	
	100 Trying ←		
	180 Ringing	← ACM	
	200 OK INVITE	← ANM	
	ACK →	7,440	
	Commu	nication	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

6.3.1.9 Call diversion (CDIV)

TP509001	SIP reference: F	RFC 3261 [6]	ES 2	ISUP reference: 283 027 [1], clause 7.4.6
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	OBCI "call diversion may	y occur" in ACM rece	eived no mapp	ing
CID Parameter	the optional backward ca procedure is not disrupte	all indicator is set to		version may occur indicator in may occur", the SIP signalling
SIP Parameter	No mapping			
values:				
ISUP Parameter	ACM optional backward	call indicator call di	version may o	occur
values:				
Comments:	SIP	S	UT	ISUP
	INVITE	→	→	IAM
			←	ACM
	180 Ringing	←	+	CPG
		Ringi	ng tone	
	200 OK INVITE	←	+	ANM
	ACK	→	_	,
	7.01	=	ersation	
	BYE		-13ali011 	REL
	· · =	(7	: :==
	200 OK BYE	→	→	RLC

TP509002	SIP reference: RF	C 3261 [6]	ES 2	ISUP reference: 283 027 [1], clause 7.4.6
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection criteria:	NOT PICS 5/12 AND NOT	PICS 5/13 AND NO	OT PICS 5/14	4 AND NOT PICS 5/15
ISUP selection criteria:				
Test purpose:	BCI called party status "no	indication" in ACM	received no	mapping
	Ensure that the SUT if a ACM is received called party status indicator "no indication" and containing a Redirection number, call diversion information, redirection number restriction and generic notification set to "Call is diverting" , the SIP signalling procedure is not disrupted (CFU, CFB, Cdi).			
SIP Parameter	No mapping			
values:				
ISUP Parameter	ACM: Redirection numbe	r, Call diversion info	rmation, Rec	lirection number restriction,
values:	Generic notification			
Comments:	SIP	SU	Т	ISUP
	INVITE	→	→	IAM
			←	ACM
	180 Ringing	←	←	CPG
	3 3	Ringing	a tone	
	200 OK INVITE	←	+	ANM
	ACK	→		
	, tort	Conver	sation	
	BYE	←	4	REL
	200 OK BYE	→	→	RLC
	ZOU ON DIE			NLO

TP509003	SIP reference: RFC 32	261 [6]		ISUP reference:
			ES 283 027	[1], clause 7.4.6
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	NOT PICS 5/12 AND NOT PIC	S 5/13 AND N	OT PICS 5/14	AND NOT PICS 5/15
criteria:				
ISUP selection				
criteria:				
Test purpose:	CPG PROGRESS with Redire		Call diversion	information and Generic
	notification received, no mapp	ing		
	Ensure that the SUT if a CPG			
				nd generic notification set to
		gnalling proced	ure is not disr	rupted (CDa, CFNR, subsequent
OID D	redirection).			
SIP Parameter	No mapping			
values:	10011 0 11 1 1 1 1 1 1			
ISUP Parameter	ACM: Called party status "Subscriber free"			
values:	CPG: Redirection number, Ca			
Comments:	SIP	SU		ISUP
	INVITE	→	→	IAM
			+	ACM
	180 Ringing	←	←	CPG
		Ringin	g tone	
	200 OK INVITE	←	←	ANM
	ACK	→		
		Conve	rsation	
	BYE	←	←	REL
	200 OK BYE	→	→	RLC

TP509004	SIP reference: RF	C 3261 [6]	_	SUP reference:
			ES 283	3 027 [1], clause 7.4.6
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	NOT PICS 5/12 AND NOT	PICS 5/13 AND NO	OT PICS 5/14 A	ND NOT PICS 5/15
criteria:				
ISUP selection				
criteria:				
Test purpose:	Redirection number restrict	tion received in AN	M no mapping	
	Ensure that the SUT if an A			number restriction
	parameter, the SIP signall	ing procedure is no	t disrupted.	
SIP Parameter	No mapping			
values:				
ISUP Parameter	ANM: Redirection number	restriction		
values:				
Comments:	SIP	SU	JT I	SUP
	INVITE	→	→	AM
			← /	ACM
	180 Ringing	←	← (CPG
		Ringin	g tone	
	200 OK INVITE	←	← /	ANM
	ACK	→		
		Conve	rsation	
	BYE	←	← 1	REL
	200 OK BYE	→	→ [RLC

TP509005	SIP reference: RFC 3261 [6]		JP reference: 127 [1], clause 7.5.4		
TSS reference:	SIP-ISUP/SS/CDIV/				
SIP selection	PICS 10/7				
criteria:					
ISUP selection					
criteria:		 			
Test purpose:	BCI called party status "no indication" in ACM	received, mappir	ng of Redirection reason.		
	Ensure that the SUT, on receipt of an ACM m	essage indicating	a first diversion with the		
	Backward call indicators parameter coded				
	Called party's status indicator = no indication	dod			
	the Call diversion information parameter co Notification subscription option = "010"B	oded			
	Redirection reason = ISUP_REASON				
	and the Generic notification indicator parai	neter coded			
	Notification indicator = call is diverting,				
	Redirection number (PIXIT) received.				
		A 181 Being Forwarded is sent. The Redirection number included in the ACM is mapped			
	into the History-Info header in the 181 Being Forwarded. A Privacy header field "history" is				
	escaped in the URI identified the diverted to user. The redirection reason is mapped into				
SIP Parameter	the cause-param in of the hi-targeted-uri identifying the diverted-to user. 181 Being Forwarded: History-Info:				
values:	hi-targeted-to-uri served user; index=1,				
values.	hi-targeted-to-un served user; index=1, hi-targeted-ti uri diverted to user; cause= Status-Code ; ?Privacy=history; index=1.1				
ISUP Parameter			,, ,		
values:					
Comments:	SIP	JT I	SUP		
	INVITE →	→	AM		
	181 Being Forwarded ←	← /	ACM		
	180 Ringing ←	← (CPG(Alerting)		
	200 OK INVITE ←	← /	MMA		
	ACK →				
	Commu				
	BYE →	-	REL		
	200 OK BYE ←	← F	RLC		
					

ISUP Parameter	Derived value of parameter field	SIP component	Value
Call diversion information			History-Info header
Redirection reason	ISUP_REASON	Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL Status-	302
	User Busy '0001'B	Code	486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509006	SIP reference: RFC 3261 [6]		ISUP reference:	
		ES 28	3 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15		
criteria:				
ISUP selection				
criteria:				
Test purpose:	BCI called party status "subscriber free" in AC	M received, m	apping of Redirection reason.	
	Ensure that the SUT on receipt of an ACM me	ssage indicatii	ng a first diversion with the	
	Backward call indicators parameter coded	J		
	Called party's status indicator = subscriber fre	е		
	the Call diversion information parameter co	ded		
	Notification subscription option = "010"B			
	Redirection reason = ISUP_REASON			
	and the Generic notification indicator parar	neter coded		
	Notification indicator = call is diverting,			
		Redirection number (PIXIT) received		
	A 180 Ringing is sent. The Redirection number included in the ACM is mapped into the			
	History-Info header in the 180 Ringing. A Privacy header field "history" is escaped in the			
	URI identified the diverted to user. The redirection reason is mapped into the cause-param in of the hi-targeted-uri identifying the diverted-to user.			
SIP Parameter	180 Ringing: History-Info:			
values:	hi-targeted-to-uri served user; index=1,			
values.	hi-targeted-to-un served user; index=1, hi-targeted-ti uri diverted to user; cause= Status-Code ; ?Privacy=history; index=1.1			
ISUP Parameter	Til-targeted-truit diverted to user; cause= Status-Code ; ?Privacy=nistory; index=1.1			
values:				
Comments:	SIP SU	ΙΤ	ISUP	
	INVITE ->	→	IAM	
	180 Ringing ←	←	ACM	
	200 OK INVITE	←	ANM	
	ACK →			
	Commu	nication		
	BYE →	→	REL	
	200 OK BYE ←	←	RLC	

ISUP Parameter	Derived value of parameter	SIP component	Value
	field		
Call diversion information			History-Info header
Redirection reason	ISUP_REASON	Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL Status-	302
	User Busy '0001'B	Code	486
	No reply '0010'B		408
	Deflection during alerting		487
	'0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509007	SIP reference: RFC 3261 [6]	ISUP reference:			
		ES 283 027 [1], clause 7.5.4			
TSS reference:	SIP-ISUP/SS/CDIV/				
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15				
criteria:					
ISUP selection					
criteria:					
Test purpose:	CPG with Event indicator ALERTING receive	d, mapping of Redirection reason.			
	Ensure that the SUT, on receipt of a CPG me	ssage indicating a first diversion with the			
	Event information parameter coded				
	Event indicator = ALERTING,				
	the Call diversion information parameter of	oded			
	Notification subscription option = "010"B				
	Redirection reason = ISUP_REASON				
	and the Generic notification indicator para	meter coded			
	Notification indicator = call is diverting,				
	Redirection number (PIXIT) received.				
	A 180 Ringing is sent. The Redirection number included in the CPG is mapped into the				
	History-Info header in the 180 Ringing. A Privacy header field "history" is escaped in the				
	URI identified the diverted to user. The redirection reason is mapped into the cause-param in of the hi-targeted-uri identifying the diverted-to user.				
SIP Parameter	180 Ringing: History-Info:				
values:	hi-targeted-to-uri served user; index=1,				
	hi-targeted-ti uri diverted to user; cause= Status-Code ; ?Privacy=history; index=1.1				
ISUP Parameter		· · · · · · · · · · · · · · · · · · ·			
values:					
Comments:	SIP SI	JT ISUP			
	INVITE ->	→ IAM			
		 ACM(no indication) 			
	180 Ringing ←	← CPG`			
	200 OK INVITE	← ANM			
	ACK →				
		nication			
	BYE →	→ REL			
	200 OK BYE ←	← RLC			
	ZOO ON DIE	I ILLO			

ISUP Parameter	Derived value of parameter field	SIP component	Value
Call diversion information			History-Info header
Redirection reason	ISUP_REASON	Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL Status-	302
	User Busy '0001'B	Code	486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509008	SIP reference: RFC 3261 [6]	ISUP reference:		
		ES 283 027 [1], clause 7.5.4		
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15		
criteria:				
ISUP selection				
criteria:				
Test purpose:	CPG with Event indicator PROGRESS received	ed, mapping of Redirection reason.		
	Ensure that the SUT, on receipt of a CPG message indicating a first diversion with the Event information parameter coded Event indicator = PROGRESS, the Call diversion information parameter coded Notification subscription option = "010"B Redirection reason = ISUP_REASON and the Generic notification indicator parameter coded Notification indicator = call is diverting, Redirection number (PIXIT) received A 181 Being Forwarded is sent. The Redirection number included in the CPG is mapped into the History-Info header in the 181 Being Forwarded. A Privacy header field "history" is			
	escaped in the URI identified the diverted to user. The redirection reason is mapped into the cause-param in of the hi-targeted-uri identifying the diverted-to user.			
SIP Parameter values:	181 Being Forwarded: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause= Status-Code ; ?Privacy=history; index=1.1			
ISUP Parameter				
values:				
Comments:	SIP SU			
	INVITE →	→ IAM		
	180 Ringing ←	← ACM		
	181 Being Forwarded ←	← CPG		
	200 OK INVITE ←	← ANM		
	ACK →			
	Commu	nication		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

ISUP Parameter	Derived value of parameter field	SIP component	Value
Call diversion information			History-Info header
Redirection reason	ISUP_REASON	Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL Status-	302
	User Busy '0001'B	Code	486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509009	SIP reference: RFC 3261 [6]	ISUP reference:		
		ES 283 027 [1], clause 7.5.4		
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	BCI called party status "no indication" in ACM received, mapping of Notification subscription option. Ensure that the SUT, on receipt of an ACM message indicating a first diversion with the			
	Backward call indicators parameter coded			
	Called party's status indicator = no indication			
	the Call diversion information parameter of			
	Notification subscription option = ISUP_N			
	Redirection reason = unconditional			
	and the Generic notification indicator parameter coded			
	Notification indicator = call is diverting,			
	Redirection number (PIXIT) received.			
	A 181 Being Forwarded is sent. The Redirection number included in the ACM is mapped			
	into the History-Info header in the 181 Being Forwarded. A Privacy header field "history" is			
OID Danamatan	escaped in the URI identified the diverted to	user.		
SIP Parameter values:	181 Being Forwarded: History-Info:			
values.	hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=302;	· 2priv value: index=1.1		
ISUP Parameter	ill-targeted-truit diverted to dser, cause=502,	, :piiv-vaiue, iiidex=1.1		
values:				
Comments:	SIP	UT ISUP		
	INVITE →	→ IAM		
	181 Being Forwarded	← ACM		
	180 Ringing	← CPG(Alerting)		
	200 OK INVITE	← ANM		
	ACK →	7 U VIVI		
	-	unication		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

	SIP component History-Info header, priv-value component	Call diversion information Notification subscription options ISUP_NSO
VA_01	Privacy header field absent or "none"	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "history"	ISUP_NSO = presentation allowed without redirection number

TP509010	SIP reference: RFC 3261 [6]	·	SUP reference: 3 027 [1], clause 7.5.4			
TSS reference:	SIP-ISUP/SS/CDIV/					
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15					
ISUP selection criteria:						
Test purpose:	BCI called party status "subscriber free" in ACM received, mapping of Notification subscription option. Ensure that the SUT on receipt of an ACM message indicating a first diversion with the Backward call indicators parameter coded Called party's status indicator = subscriber free the Call diversion information parameter coded Notification subscription option = ISUP_NSO Redirection reason = unconditional and the Generic notification indicator parameter coded Notification indicator = call is diverting, Redirection number (PIXIT) received. A 180 Ringing is sent. The Redirection number included in the ACM is mapped into the History-Info header in the 180 Ringing. A Privacy header field "history" is escaped in the URI identified the diverted to user.					
SIP Parameter values:	180 Ringing: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=302; ?priv-value; index=1.1					
ISUP Parameter values:	ANM: Redirection number restriction					
Comments:	SIP SUT ISUP					
	INVITE →	→	IAM			
	180 Ringing ←	←	ACM			
	200 OK INVITE ←	←	ANM			
	ACK →	-!!!				
	Commu		DEL			
	- · -	BYE → REL				
	200 OK BYE ←	<u> </u>	RLC			

	SIP component History-Info header, priv-value component	Call diversion information Notification subscription options ISUP_NSO
VA_01		ISUP_NSO = presentation allowed with redirection number
VA 02		ISUP_NSO = presentation allowed without
VA_02	, ,	redirection number

TP509011	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4				
TSS reference:	SIP-ISUP/SS/CDIV/					
SIP selection		PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15				
criteria:						
ISUP selection						
criteria:						
Test purpose:	CPG with Event indicator ALERTING receiv	ed, mapping of Notification subscription option.				
	Ensure that the SUT on receipt of a CPG me	essage indicating a first diversion with the				
	Event information parameter coded	3 3				
	Event indicator = ALERTING,					
	the Call diversion information parameter					
	Notification subscription option = ISUP_NSO)				
	Redirection reason = unconditional					
	and the Generic notification indicator par	ameter coded				
	Notification indicator = call is diverting,					
	Redirection number (PIXIT) received.					
	A 180 Ringing is sent. The Redirection number included in the CPG is mapped into the					
	URI identified the diverted to user.	ivacy header field "history" is escaped in the				
SIP Parameter	180 Ringing: History-Info:					
values:	hi-targeted-to-uri served user; index=1,					
	hi-targeted-ti uri diverted to user; cause=302	?; ?priv-value; index=1.1				
ISUP Parameter						
values:						
Comments:	SIP	SUT ISUP				
	INVITE →	→ IAM				
		← ACM				
	180 Ringing ←	← CPG				
	200 OK INVITE ←	← ANM				
	ACK →					
	Comm	unication				
	BYE →	→ REL				
	200 OK BYE ←	← RLC				

	SIP component History-Info header, priv-value component	Call diversion information Notification subscription options ISUP_NSO
VA_01	Privacy header field absent or "none"	ISUP_NSO = presentation allowed with
		redirection number
VA_02	Privacy "history"	ISUP_NSO = presentation allowed without
		redirection number

TP509012	SIP reference: RFC 326	1 [6]		ISUP reference: 3 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
ISUP selection criteria:				
Test purpose:	CPG with Event indicator ALER	TING received	l, mapping of F	Redirection number restriction.
	Ensure that the SUT on receipt of a CPG message indicating a first diversion with the Event information parameter coded Event indicator = ALERTING, Redirection number restriction parameter = ISUP_RDIR_RESTR A 180 Ringing including a History-Info header is sent. A Privacy header field "history" is escaped in the URI identified the diverted to user.			
SIP Parameter	180 Ringing: History-Info:			
values:	hi-targeted-to-uri served user; inc			
IOUD D	hi-targeted-ti uri diverted to user;		?priv-value; ind	dex=1.1
ISUP Parameter values:	ANM: Redirection number restri	ction		
Comments:	SIP	SU	T .	ISUP
	- · ·	→	•	IAM
		- -	-	ACM
		- -	÷	CPG
	0 0	E	-	ANM
		→	_	
	Communication			
	BYE -	→	→	REL
	200 OK BYE	-	+	RLC

	Derived escaped SIP priv-value	Derived value of parameter field	
	component		
VA_01	Privacy header field absent or "none"	ISUP_RDIR_RESTR = Presentation allowed, '00'B	
VA_02	Privacy header field "history" and "id"	ISUP_RDIR_RESTR = presentation restricted, '01'B	
VA_03	Privacy header field absent or "none"	ISUP_RDIR_RESTR absent	

TP509013	SIP reference: RFC 3	261 [6]		ISUP reference: 3 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
ISUP selection criteria:				
Test purpose:	ANM received, mapping of the	e Redirection n	umber restrictio	n parameter.
	Ensure that the SUT, on receipt of an ANM message with the Redirection number restriction parameter = ISUP_RDIR_RESTR or parameter absent, a 200 IK INVITE including a History-Info header is sent. A Privacy header field is escaped in the URI identified the diverted to user according the value of the Redirection number restriction parameter.			
SIP Parameter	200 OK INVITE: History-Info:			
values:	hi-targeted-to-uri served user; hi-targeted-ti uri diverted to us		; ?priv-value; inc	dex=1.1
ISUP Parameter	ANM: Redirection number re-			
values:				
Comments:	SIP	S	UT	ISUP
	INVITE	→	→	IAM
	181 Being Forwarded	←	←	ACM
	180 Ringing	←	←	CPG
	200 OK INVITE	←	←	ANM
	ACK	→		
		Comm	unication	
	BYE	→	→	REL
	200 OK BYE	←	+	RLC
Comments:	SIP	S	UT	ISUP
	INVITE	→	→	IAM
	180 Ringing	←	←	ACM
	200 OK INVITE	←	←	ANM
	ACK	→		
		Comm	unication	
	BYE	→	→	REL
	200 OK BYE	←	+	RLC

	Derived escaped SIP component	Derived value of parameter field
VA_01	Privacy header field absent or "none"	ISUP_RDIR_RESTR = Presentation allowed, '00'B
VA_02	Privacy header field "history" and "id"	ISUP_RDIR_RESTR = presentation restricted, '01'B
VA 03	Privacy header field absent or "none"	ISUP RDIR RESTR absent

TP509014	SIP reference: RFC 3261 [6]		SUP reference: 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/ CDIV /	E3 203	027 [1], Clause 7.5.4
SIP selection	31F-130F/33/ CDIV /		
criteria:			
ISUP selection	NOT PICS 1/5 AND PICS 10/6		
criteria:			
Test purpose:	CDIV performed, the first hi-targeted-to-uri is sent in the original called number national number		
	Ensure that the SUT in the Idle state Sends an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, Privacy header field is absent and with the complete Original called number parameter contained in the URI of first Index entry of History-Info header in the format "+" CC NDC SN.		
	of a IAM message with the Redirection information parameter coded Redirection counter = 1 Redirecting reason = ISUP_RR and the Original called number parameter coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed		
SIP Parameter	Address signals included in the format CC+NDC+SN. INVITE: History-Info: hi-targeted-to-uri served user; index=1,		
values:	hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1		
ISUP Parameter	IAM: Original called number parameter code	ed	
values:	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony n		
	Address presentation restricted parameter = presentation allowed		
Comments:	Address signals userinfo of the hi-targeted-to from index 1 SIP SUT ISUP		
Comments:	SIP SU	·	ISUP IAM
		7	ACM
	180 Ringing ← 200 OK INVITE ←	-	ANM
	ACK +	•	VINIAI
	Commu	nication	
	BYE →	→	REL
	200 OK BYE ←	-	RLC

ISUP Parameter	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value
	unknown '0000'B	_"cause" EQUAL	404
	Unconditional '0011'B	Status-Code	302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509015	SIP reference: RFC 3261 [6]		SUP reference: 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/ CDIV /	E3 203	027 [1], clause 7.5.4
SIP selection	5IP-15UP/55/ CDIV /		
criteria:			
ISUP selection	PICS 1/5 AND PICS 10/6		
criteria:	1100 1/0 / 1100 10/0		
Test purpose:	CDIV performed, the first hi-targeted-to-uri is sent in the original called number international number		
	Ensure that the SUT in the Idle state, on receipt an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, the Privacy header is ansent and with the complete Original called number parameter contained URI of first Index entry of History-Info header in the format "+" CC NDC SN.		
	Sends of a IAM message with the Redirect	ction informati	on parameter coded
	Redirection counter = 1		
	Redirecting reason = ISUP_RR		
	and the Original called number parameter coded		
	Nature of address indicator = international number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address presentation restricted parameter = presentation allowed Address signals included in the format CC+NDC+SN		
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served user; index=1,		
values:	hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1		
ISUP Parameter	IAM: Original called number parameter coded		
values:	Nature of address indicator = international nur		
	Numbering plan indicator = ISDN/Telephony n		
	Address presentation restricted parameter = presentation allowed		
	Address signals userinfo of the hi-targeted-to from index 1		
Comments:	SIP SU	ΙΤ	ISUP
	INVITE →	→	IAM
	180 Ringing ←	←	ACM
	200 OK INVITE ←	+	ANM
	ACK →		
	Commu	nication	
	BYE →	→	REL
	200 OK BYE ←	+	RLC

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value
	unknown '0000'B	"cause" EQUAL	404
	Unconditional '0011'B	Status-Code	302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during		487
	alerting '0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509016	SIP reference: RFC 3261 [6]		SUP reference:
TSS reference:	CID ICUD/CC/ CDIV/	E3 203	027 [1], clause 7.5.4
SIP selection	SIP-ISUP/SS/ CDIV /		
criteria:			
ISUP selection	PICS 10/6		
criteria:	1 103 10/0		
Test purpose:	CDIV performed, the first hi-targeted-to-uri is sent in the original called number Privacy header is equal "history"		
	Ensure that the SUT in the Idle state, INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, the priv-value set to "history" and with the complete Original called number parameter contained URI of first Index entry of History-Info header in the format "+" CC NDC SN.		
	Sends an a IAM message with the Redirection information parameter coded Redirection counter = 1 Redirecting reason = ISUP_RR		
	and the Original called number parameter coded		
	Address presentation restricted parameter = presentation restricted		
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1,		
values:	hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1		
ISUP Parameter	IAM: Original called number parameter coded		
values:	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address presentation restricted parameter = presentation restricted Address signals userinfo of the hi-targeted-to from index 1		
Comments:	SIP SUT ISUP		
	INVITE →	→	IAM
	180 Ringing	-	ACM
	200 OK INVITE	+	ANM
	ACK →	-	
	Communication		
	BYE →		REL
	200 OK BYE ←	+	RLC

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value
	unknown '0000'B	"cause" EQUAL	404
	Unconditional '0011'B	Status-Code	302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509017	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /	E3 203 027 [1], Clause 7.3.4	
SIP selection			
criteria:			
ISUP selection	NOT PICS 1/5 AND PICS 10/6		
criteria:	11011100 1/0711101100 10/0		
Test purpose:	CDIV performed, the first hi-targeted-to-uri is sent in the redirecting number national number		
	Ensure that the SUT in the Idle state Sends an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, Privacy header field is absent and with the complete Redirecting number parameter contained in the hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN.		
	of a IAM message with the Redirection inform	mation parameter coded	
	Redirection counter = 1		
	Redirecting reason = ISUP_RR		
	and the Redirecting number parameter coded		
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address presentation restricted parameter = presentation allowed		
OID D	Address signals included in the format CC+NDC+SN		
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served user; index=1,		
values:	hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1		
ISUP Parameter	IAM: Redirecting number parameter coded		
values:	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony n		
	Address presentation restricted parameter = presentation allowed		
Comments:	Address signals userinfo of the hi-targeted-to from index 1 SIP SUT ISUP		
Comments.	INVITE →	→ IAM	
		← ACM	
	180 Ringing ← 200 OK INVITE ←	← ACM ← ANM	
		▼ AINIVI	
	ACK → Commu	nication	
	BYE -	→ REL	
	200 OK BYE ←	← RLC	

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value
	unknown '0000'B	"cause" EQUAL	404
	Unconditional '0011'B	Status-Code	302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during		487
	alerting '0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509018	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /	203 027 [1], clause 7.5.4	
SIP selection	SIF-ISOF/SS/ CDIV /		
criteria:			
ISUP selection	PICS 1/5 AND PICS 10/6		
criteria:			
Test purpose:	CDIV performed, the first hi-targeted-to-uri is sent in the redirecting number international number		
	Ensure that the SUT in the Idle state, on receipt an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, Privacy header field is absent and with the complete Redirecting number parameter contained hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN.		
	Sends of a IAM message with the Redirection	n information parameter coded	
	Redirection counter = 1		
	Redirecting reason = ISUP_RR		
	and the Redirecting number parameter coded		
	Nature of address indicator = international number		
	Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed		
	Address signals included in the format CC+NDC+SN		
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served user; index=1,		
values:	hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1		
ISUP Parameter	IAM: Redirecting number parameter coded		
values:	Nature of address indicator = international nur	nber	
	Numbering plan indicator = ISDN/Telephony n		
	Address presentation restricted parameter = presentation allowed		
	Address signals userinfo of the hi-targeted-to from index 1		
Comments:	SIP SUT ISUP		
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	200 OK INVITE ←	← ANM	
	ACK →		
	Commu		
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value
	unknown '0000'B	"cause" EQUAL	404
	Unconditional '0011'B	Status-Code	302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during		487
	alerting '0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509019	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /	L3 203 027 [1], clause 7.3.4	
SIP selection	311 -1301 /33/ CDIV /		
criteria:			
ISUP selection	PICS 10/6		
criteria:			
Test purpose:	CDIV performed, the second hi-targeted-to-uri is sent in the redirecting number Privacy header is equal "history"		
	Ensure that the SUT in the Idle state, INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, the priv-value set to "history" and with the complete Redirecting number parameter contained hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN.		
	Sends an a IAM message with the Redirection information parameter coded Redirection counter = 1 Redirecting reason = ISUP_RR		
	and the Redirecting number parameter coded		
OID D	Address presentation restricted parameter = presentation restricted		
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1,		
ISUP Parameter	hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1 IAM: Redirecting number parameter coded		
values:	Nature of address indicator = international nur	nher	
values.	Numbering plan indicator = ISDN/Telephony n		
	Address presentation restricted parameter = p		
	Address signals userinfo of the hi-targeted-to from index 1		
Comments:	SIP SU	T ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	200 OK INVITE ←	← ANM	
	ACK →		
	Commu		
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value
	unknown '0000'B Unconditional '0011'B	"cause" EQUAL Status-Code	302
	User Busy '0001'B No reply '0010'B		486 408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509020	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /		
SIP selection			
criteria:			
ISUP selection	PICS 10/		
criteria:			
Test purpose:	CDIV performed, the second hi-targeted-to-uri Privacy header is not included is sent in the redirecting number and the first hi-targeted-to-uri Privacy header is sent in the original called number Ensure that the SUT in the Idle state, on receipt of an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table and with the complete Original called number parameter contained the URI of first Index entry of History-Info header, the Privacy header field is absent in the format "+" CC NDC SN. The Redirecting number parameter is contained in the second hi-targeted-to-uri of		
SIP Parameter values:	History-Info header in the format "+" CC NDC SN the Privacy header field is absent. Sends a IAM message with the Redirection information parameter coded Redirection counter 2 Redirecting reason = ISUP_RR, the Original called number parameter coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included and the Redirecting number parameter coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user C; cause=302; index=1.1		
IOUD Danson of an	hi-targeted-ti uri diverted to user D; cause=Status-Code; index=1.1.1		
ISUP Parameter values:	IAM: Original called number parameter cod Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1 Redirecting number parameter coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1.1		
Comments:	SIP SU		
	INVITE -	→ IAM	
	180 Ringing	← ACM	
	200 OK INVITE ← ACK →	← ANM	
	-	nication	
	Commu		
	BYE -	→ REL	
	200 OK BYE ←	← RLC	

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value
	unknown '0000'B	"cause" EQUAL	404
	Unconditional '0011'B	Status-Code	302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509021	SIP reference: RFC 3261 [6]	ISUP reference:	
TSS reference:	SIP-ISUP/SS/ CDIV /	ES 283 027 [1], clause 7.5.4	
SIP selection	31F-130F/33/ CDIV /		
criteria:			
ISUP selection	PICS 10/6		
criteria:	1100 10/0		
Test purpose:	CDIV performed, the second hi-targeted-to-ur	i Privacy ="history" is sent in the redirecting	
	number and the first hi-targeted-to-uri Privacy		
	,	· ·	
	Ensure that the SUT in the Idle state, on receipt of an INVITE message with Cause Value		
	in History Index; cause-param = "cause" EQUAL Status-Code defined in the table and with		
	the complete Original called number parametric before the Driver when the Driv		
	History-Info header, the Privacy header field is The Redirecting number parameter is conta		
	History-Info header in the format "+" CC NDC		
	I listory-line header in the format + CC NDC	Siv, the i fivacy value is set to flistory .	
	Sends a IAM message with the Redirection i	nformation parameter coded	
	Redirection counter 2		
	Redirecting reason = ISUP_RR,		
	the Original called number parameter code		
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony r		
	Address presentation restricted parameter = presentation allowed		
	Address signals included and the Redirecting number parameter code	ad	
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address presentation restricted parameter = presentation restricted		
	Address signals included		
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served user; index=1,		
values:		to user C?Privacy=history; cause=302;	
	index=1.1		
IOUD Deservator		to user D; cause=Status-Code; index=1.1.1	
ISUP Parameter values:	IAM: Original called number parameter cod Nature of address indicator = national number		
values.	Numbering plan indicator = ISDN/Telephony r		
	Address presentation restricted parameter = p		
	Address signals userinfo of the hi-targeted-to		
	Redirecting number parameter coded		
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony r	- ·	
	Address presentation restricted parameter = presentation restricted		
0	Address signals userinfo of the hi-targeted-to		
Comments:	SIP SU		
	INVITE 180 Ringing ←	→ IAM	
	1.00.1	← ACM	
	200 OK INVITE ← ACK →	← ANM	
	ACK → Commu	nication	
	BYE →	→ REL	
		→ REL ← RLC	
	200 OK BYE ←	₹ KLU	

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information		Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL	302
	User Busy '0001'B	Status-Code	486
	No reply '0010'B		408
	Deflection during		487
	alerting '0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509022	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /		
SIP selection			
criteria:			
ISUP selection	PICS 10/6		
criteria:			
Test purpose:	CDIV performed, the second hi-targeted-to-uri Privacy header absent is sent in the redirecting number and the first hi-targeted-to-uri Privacy = "history" is sent in the original called number Privacy Ensure that the SUT in the Idle state, on receipt of an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table and with the complete Original called number parameter contained the URI of first Index entry of History-Info header, the Privacy value set to "history" in the format "+" CC NDC SN. The Redirecting number parameter is contained in the second hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN the Privacy header field is absent. Sends an IAM message with the Redirection information parameter coded Redirection counter 2 Redirecting reason = ISUP_RR,		
	the Original called number parameter coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals included and the Redirecting number parameter coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included		
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1, hi-targeted-ti uri diverted to user C; cause=302; index=1.1 hi-targeted-ti uri diverted to user D; cause=Status-Code; index=1.1.1		
ISUP Parameter	IAM: Original called number parameter cod		
values:	Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address userinfo of the hi-targeted-to from index 1 Redirecting number parameter coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1.1		
Comments:	SIP SU		
	INVITE -	→ IAM	
	180 Ringing	← ACM	
	200 OK INVITE	← ANM	
	ACK →	and a matter of	
	Commu		
	BYE -	→ REL	
	200 OK BYE ←	← RLC	

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information		Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL	302
	User Busy '0001'B	Status-Code	486
	No reply '0010'B		408
	Deflection during		487
	alerting '0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509023	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /		
SIP selection			
criteria:			
ISUP selection	PICS 10/6		
criteria:			
Test purpose:	CDIV performed, the second hi-targeted-to-uri Privacy="history" is sent in the redirecting number and the first hi-targeted-to-uri Privacy = "history" is sent in the original called number Privacy Ensure that the SUT in the Idle state, on receipt of an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table and with the complete Original called number parameter contained the URI of first Index entry of History-Info header the Privacy is set to "history" in the format "+" CC NDC SN.		
SIP Parameter	The Redirecting number parameter is contained in the second hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN the Privacy is set to "history". Sends an IAM message with the Redirection information parameter coded Redirection counter 2 Redirection reason = ISUP_RR, the Original called number parameter coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals included and the Redirecting number parameter coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address presentation restricted parameter = presentation restricted Address signals included INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1,		
values:	hi-targeted-ti uri diverted to user C?Privacy=history; cause=302; index=1.1		
	hi-targeted-ti uri diverted to user D; cause=Status-Code; index=1.1.1		
ISUP Parameter	IAM: Original called number parameter cod		
values:	Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals userinfo of the hi-targeted-to from index 1 Redirecting number parameter coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals userinfo of the hi-targeted-to from index 1.1		
Comments:	SIP SU		
	INVITE ->	→ IAM	
	180 Ringing ←	← ACM	
	200 OK INVITE ←	← ANM	
	ACK →		
	Commu		
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

ISUP Parameter or IE	Derived value of	SIP component	Value
	parameter field		
IAM		INVITE	
Redirection Information		Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL	302
	User Busy '0001'B	Status-Code	486
	No reply '0010'B		408
	Deflection during		487
	alerting '0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509024	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /		
SIP selection			
criteria:			
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN		
Test purpose:	CDIV performed, the third hi-targeted-to-uri is sent in the redirecting Redirection counter is mapped from the latest history entry		
	Ensure that the SUT in the Idle state, on receipt of an INVITE message containing a History-Info header with three History entries, the hi-targeted-to-uri of first index is mapped into the Original called number parameter ; the hi-targeted-to-uri of third index is mapped into the Redirecting number parameter Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table.		
	Sends a IAM message with the Redirection information parameter coded Redirection counter 3		
	Redirecting reason = ISUP_RR, the Original called number parameter code	4	
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address signals included		
	and the Redirecting number parameter coded		
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
SIP Parameter	Address signals included INVITE: History-Info: hi-targeted-to uri served user; index=1,		
values:	hi-targeted-ti uri diverted to user C; cause=302index=1.1		
Valuoo.	hi-targeted-ti uri diverted to user D; cause=486index=1.1.1		
	hi-targeted-ti uri diverted to user E; cause=Status-Code; index=1.1.1.1		
ISUP Parameter	IAM: Original called number parameter coded		
values:	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address signals userinfo of the hi-targeted-to from index 1 Redirecting number parameter coded		
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address signals userinfo of the hi-targeted-to from index 1.1.1		
Comments:	SIP SU	JT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	200 OK INVITE ←	← ANM	
	ACK →		
	Commu		
	BYE -	→ REL	
	200 OK BYE ←	← RLC	

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information		Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL	302
	User Busy '0001'B	Status-Code	486
	No reply '0010'B		408
	Deflection during		487
	alerting '0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509025	SIP reference: RFC 3261 [6]		ISUP reference: 3 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/ CDIV /			
SIP selection criteria:				
ISUP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/1	3 AND NC	T PICS 5/14 /	AND NOT PICS 5/15
Test purpose:	Interworking not supported, session	successfu	l.	
	Ensure that the SUT in the Idle state, on receipt of an INVITE message containing a History-Info header with three History entries the History-Info header entries are not mapped into any call diversion relateded parameters in the IAM and the session setup is not disruppted.			
SIP Parameter	INVITE: History-Info: hi-targeted-to-u			
values:				se=Status code; index=1.1 se=Status code; index=1.1.1
ISUP Parameter	IAM: no mapping			
values:				
Comments:	SIP	SU		ISUP
	INVITE →		→	IAM
	180 Ringing ←		←	ACM
	200 OK INVITE		←	ANM
	ACK →			
		Commun	ication	
	BYE →		→	REL
	200 OK BYE ←		<u> </u>	RLC

6.3.1.10 Call waiting (CW)

TP510001	SIP reference: F	RFC 3261 [6]	ES 2	ISUP reference: 283 027 [1], clause 7.4.9
TSS reference:	SIP-ISUP/SS/CW/			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	No mapping of Call Wait Ensure that the indication a waiting call" is not inte	n for Call Waiting co	ontained in an	ACM, Generic notification "call is
SIP Parameter values:			-	
ISUP Parameter values:	ACM: Generic notification	on parameter = "Ca	ll is a waiting o	all"
Comments:	SIP	9	SUT	ISUP
	INVITE	→	→	IAM
	180 Ringing	← Ring	← ing tone	ACM
	200 OK INVITE	← ~ ~	<u> </u> ←	ANM
	ACK	→		
		Conv	ersation	
	BYE	←	(REL
	200 OK BYE	→	→	RLC

TP510002	SIP reference: RFC 3261 [6]	Ee	ISUP reference: 283 027 [1], clause 7.4.9
TSS reference:	SIP-ISUP/SS/CW/	L ES	263 027 [1], Clause 7.4.9
SIP selection	31F-130F/33/CW/		
criteria:			
ISUP selection			
criteria:			
Test purpose:	No mapping of Call Waiting indication	in the CPG at the I	-MGCF
	Ensure that the indication for Call Wait a waiting call", is not interworked in a		
SIP Parameter	180 Ringing		
values:			
ISUP Parameter	ACM: Called party status "no indication	n"	
values:	CPG: Generic notification parameter	= "Call is a waiting	call"
Comments:	SIP	SUT	ISUP
	INVITE →	-	IAM
		•	- ACM
	180 Ringing ←	•	- CPG
		Ringing tone	
	200 OK INVITE ←	•	- ANM
	ACK →		
		Conversation	
	BYE ←	•	REL
	200 OK BYE →		RLC

6.3.1.11 User to user signalling (UUS)

TP511001	SIP reference: RFC 3261 [6]		ISUP reference:
			ITU-T Red	C Q.1912.5 [32], annex B.21
			ITU-T Rec	Q.737.1 [33], clause 1.3.7.2
TSS reference:	SIP-ISUP/SS/UUS/			
SIP selection				
criteria:				
ISUP selection	PICS 11/1 AND PICS 11/2			
criteria:				
Test purpose:	Explicit request supported, a FAR L	iser-to-use	er service 3 re	quest (not essential) is rejected
	with FRJ Ensure that the SUT if a FAR is rec	aivad with	on user to u	cor corving 2 request (not
	essential) after call setup, sent a F			
	is not disrupted.	ito to rejec	or the request.	The Oil Signaming procedure
SIP Parameter	io not dioraptodi			
values:				
ISUP Parameter	FRJ: User-to-user indicator = "Se	vice 3 not	provided"	
values:				
Comments:	SIP	SL	JT	ISUP
	INVITE →		→	IAM
	180 Ringing ←		←	ACM
		Ringin		
	200 OK INVITE ←		←	ANM
	ACK →			
		Conve	rsation	
			(FAR
		_	→	FRJ
		Conve	rsation _	
	BYE +		←	REL
	200 OK BYE →		<u>→</u>	RLC

TP511002	SIP reference: RFC	3261 [6]		IT	ISUP reference: c Q.1912.5 [32], annex B.21 U-T Rec Q.737 [33], clause 1.3.5.2.5.2
TSS reference:	SIP-ISUP/SS/UUS/				
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 11/2				
Test purpose:	Explicit request not supported, no response on receipt of a FAR Ensure that the SUT if a FAR is received with an user-to-user service 3 request (not essential) after call setup, the SIP signalling procedure is not disrupted.				
SIP Parameter					
values:					
ISUP Parameter					
values:					
Comments:	SIP		SUT		ISUP
	INVITE	→		→	IAM
	180 Ringing	←		←	ACM
			Ringing tone	:	
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation	ı	
				←	FAR
			Conversation	ı	
	BYE	←		←	REL
	200 OK BYE	→		→	RLC

TP511003	SIP reference: RFC	3261 [6]	ITU	ISUP reference: c Q.1912.5 [32], annex B.21 I-T Rec Q.737.1 [33], clause 1.3.5.2.5.2
TSS reference:	SIP-ISUP/SS/UUS/			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	User-to-user service 1 implications Ensure that the SUT if an Use parameter is sent in the IAM, the uuidata component of the	er-to-User head The data field	of the User-to-	user information is derived from
SIP Parameter values:	INVITE: User-to-User uuidata			
ISUP Parameter values:	IAM: User-to-user informatio	n		
Comments:	SIP	S	UT	ISUP
	INVITE	→	→	IAM
	180 Ringing	← Ringi	← ng tone	ACM
	200 OK INVITE ACK	← →	+	ANM
	BYE	← Conv	ersation	REL
	200 OK BYE	→	→	RLC

TP511004	SIP reference: RFC 32	.61 [6 <u>]</u>	ITU-	IT	ISUP reference: c Q.1912.5 [32], annex B.21 U-T Rec Q.737 [33], lause 1.3.5.2.5.2.1
TSS reference:	SIP-ISUP/SS/UUS/				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	User-to-user service 1 respons	e			
	Ensure that the User-to-user in uuidata of a nUser-to-User hea				CM, ANM or REL is mapped in esponse, final response or BYE
SIP Parameter	18x: User-to-User uuidata				
values:	200: User-to-User uuidata				
	BYE: User-to-User uuidata				
ISUP Parameter	ACM: User-to-user information				
values:	ANM: User-to-user information				
	REL: User-to-user information				
Comments:	SIP		SUT		ISUP
	INVITE	→			IAM
	180 Ringing	←		←	ACM
			Ringing tone		
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	←		←	REL
	200 OK BYE	→		→	RLC

6.3.1.12 Explicit call transfer (ECT)

TP512001	SIP reference: RFC 3261 [6]		ES 2	ISUP reference: 83 027 [1], clause 7.4.8
TSS reference:	SIP-ISUP/SS/ECT/		L3 Z	.05 027 [1], clause 7.4.0
SIP selection	31F-130F/33/EC1/			
criteria:				
ISUP selection	PICS 12/1			
criteria:	FIGS 12/1			
Test purpose:	Loop prevention procedure supported, a	I OP respon	se "in	sufficient information" is sent
rest purpose.	Loop prevention procedure supported, a	LOT Tespons	30 111	samelent information is sent
	Ensure that the SUT if a LOP(request) is indication "insufficient information" continuous procedure. Ensure that the SUT if a FAC is received procedure.	nue without d	isrupt	ing the SIP signalling
SIP Parameter				
values:				
ISUP Parameter	LOP: Response "insufficient information	n"		
values:				
Comments:	SIP	SUT		ISUP
	INVITE →		→	IAM
	180 Ringing ←		←	ACM
	R	linging tone		
	200 OK INVITE ←		←	ANM
	ACK →			
	C	onversation		
			←	LOP
			→	LOP
			←	FAC
	C	onversation		
	BYE ←		←	REL
	200 OK BYE →		→	RLC

TP512002	SIP reference: RFC 320	61 [6]	ES 2	ISUP reference: 83 027 [1], clause 7.4.8
TSS reference:	SIP-ISUP/SS/ECT/			
SIP selection criteria:				
ISUP selection criteria:	NO PICS 12/1			
Test purpose:	Loop prevention procedure not		ivad aantinua	with out diamenting the CID
	Ensure that the SUT if a LOP(re signalling procedure. Ensure that the SUT if a FAC is procedure.			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	SIP	SU	IT	ISUP
	INVITE	→	→	IAM
	180 Ringing	←	←	ACM
		Ringin		
	200 OK INVITE	←	←	ANM
	ACK	→		
		Conver		1.00
			+	LOP
		_	+	FAC
	DVE	Conver		DEL
	BYE	((REL
	200 OK BYE	→	→	RLC

6.3.1.13 Completion of Call to Busy Subscriber (CCBS)

TP513001	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.4.11			
TSS reference:	SIP-ISUP/SS/CCBS/				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	CCBS possible in the Diagnostics field red				
	Ensure that the SUT if a REL is received of	ontained a Diagnostic field and the CCBS			
	indicator is coded as CCBS possible:				
	 continue without disrupting the SIP signalling procedure. 				
SIP Parameter					
values:					
ISUP Parameter	REL: Cause indicator Diagnostics CCBS p	ossible			
values:					
Comments:	SIP	SUT ISUP			
	INVITE →	→ IAM			
	486 Busy Here ←	← REL			
	ACK →	→ RLC			

6.3.1.14 Completion of Calls on No reply (CCNR)

TP514001	SIP reference: F	RFC 3261 [6]		ISUP reference:
			ES 2	83 027 [1], clause 7.4.12
TSS reference:	SIP-ISUP/SS/CCNR/			
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	CCNR possible in the A	CM received		
	Ensure that the SUT if a	ACM is received an	d a CCNR Po	ssible Indicator is included:
	 continue without 	ut disrupting the SIP	signalling pro	cedure.
SIP Parameter				
values:				
ISUP Parameter	ACM: CCNR possible in	dicator CCNR possib	ole	
values:				
Comments:	SIP	S	UT	ISUP
	INVITE	→	→	IAM
	180 Ringing	←	(ACM
		Ringii	ng tone	
	200 OK INVITE	←	· ←	ANM
	ACK	→		
		Conve	ersation	
	BYE	←	(REL
	200 OK BYE	→	→	RLC

6.3.1.15 Anonymous Call Rejection (ACR)

TP515001	SIP reference:	RFC 3261 [6]	ES 28	ISUP reference: 33 027 [1], clause 7.4.23
TSS reference:	SIP-ISUP/SS/ACR/			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:		a destination user ha ot is rejected with a RI		ne ACR supplementary service:
SIP Parameter	INVITE: Privacy-head	der = "id"		
values:				nendation Q.850 [5]; cause=24
ISUP Parameter values:	REL: Cause	value: 24 "call rejecte	ed due to ACR	supplementary service"
Comments:	SIP	S	UT	ISUP
	INVITE	→	→	IAM
	603 Decline	←	←	REL
	ACK	→	→	RLC

6.3.1.16 Closed user group (CUG)

TP516001	SIP reference: RFC 326	1 [6]		ISUP reference:
T00 == (================================				7.4.1.1
TSS reference:	SIP-ISUP/SS/CUG/			
SIP selection				
criteria:	DIGG 5/7			
ISUP selection criteria:	PICS 5/7			
Test purpose:	Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "00"</cug>			
	Ensure that the <cugcommunicationindicator> value "00" contained in the INVITE <cug> XML body is sent in a optional forward call indicator - CUG call indicator, if any other value of the optional forward call indicator have to be set not equal "0". No mapping of <networkindicator> and <cuginterlockbinarycode> into Closed User Group interlock code.</cuginterlockbinarycode></networkindicator></cug></cugcommunicationindicator>			
SIP Parameter	INVITE:			
values:	<cug></cug>			
	<networkindicator>[PIXIT]</networkindicator> <cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode> <cugcommunicationindicator>00</cugcommunicationindicator>			
ISUP Parameter	IAM:			
values:	Optional Forward Call Indicator CUG call indicator = "00" When optional forward call indicator have to be sent in case of an other indicator is not set to "0"			
Comments:	SIP	SU	Т	ISUP
	INVITE -	→	→	IAM
	180 Ringing	E	←	ACM
		Ringing	g tone	
	200 OK INVITE	+	←	ANM
	ACK -	→		
		Conver	sation	
	BYE	F	←	REL
	200 OK BYE	}	→	RLC

TP516002	SIP reference: RFC 3261 [6]	ISUP reference: 7.4.1.1	
TSS reference:	SIP-ISUP/SS/CUG/	•	
SIP selection			
criteria:			
ISUP selection	PICS 5/7		
criteria:			
Test purpose:	Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "01"</cug>		
	XML body is not sent in a optional forward of call indicator have to be sent, the C mapping of <networkindicator> and <cugint code.<="" interlock="" th=""><th>or> value "01" contained in the INVITE <cug> all indicator - CUG call indicator. If the optional UG call indicator is set to "00" no CUG call. No erlockBinaryCode> into Closed User Group</cug></th></cugint></networkindicator>	or> value "01" contained in the INVITE <cug> all indicator - CUG call indicator. If the optional UG call indicator is set to "00" no CUG call. No erlockBinaryCode> into Closed User Group</cug>	
SIP Parameter	INVITE:		
values:	<pre><cug> <networkindicator>[PIXIT]</networkindicator></cug></pre>		
	<pre><cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode>[PIXIT]</pre>		
ISUP Parameter	IAM:		
values:	Optional Forward Call Indicator CUG call ind When optional forward call indicator have to to "0"	dicator = "00" be sent in case of an other indicator is not set	
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Ring	ing tone	
	200 OK INVITE ←	← ANM	
	ACK →		
		rersation	
	BYE ←	← REL	
	200 OK BYE →	→ RLC	

TP516003	SIP reference: RFC 3261 [6]	ES 28	ISUP reference: 3 027 [1], clause 7.4.1.1
TSS reference:	SIP-ISUP/SS/CUG/	•	
SIP selection			
criteria:			
ISUP selection	PICS 5/7		
criteria:			
Test purpose:	Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "10"</cug>		
	Ensure that the <cugcommunicationindicator a="" body="" call="" forward="" in="" indecug="" is="" optional="" sent="" xml=""> <networkindicator> is mapped into the</networkindicator></cugcommunicationindicator>	dicator - CUG	call indicator ="10". The XML
	Network indentity and the XML <cug> <cug binar<="" closed="" code="" group="" iam="" interlock="" th="" user=""><th></th><th>aryCode> is mapped into the</th></cug></cug>		aryCode> is mapped into the
SIP Parameter	INVITE:		
values:	<cug></cug>		
	<networkindicator>[PIXIT]</networkindicator>		
	<pre><cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode> <cugcommunicationindicator>10</cugcommunicationindicator></pre>		
		mmunicationii	Idicator>
ISUP Parameter	IAM:		
values:	Optional Forward Call Indicator CUG call indicator = "10"		
	Closed User Group interlock code		
	Binary code derived from INVITE XML body < cugInterlockBinaryCode>		
	Network indentity derived from INVITE		
Comments:		JT _	ISUP
	INVITE -	→	IAM
	180 Ringing ←	+	ACM
		ig tone	ANM
	200 OK INVITE ← ACK →	~	AINIVI
	7.0.1	rsation	
	BYE	+ (15alion	REL
	200 OK BYE →	→	RLC
L	200 01(512		

TP516004	SIP reference: RFC 3261 [6]	ES 28	ISUP reference: 3 027 [1], clause 7.4.1.1
TSS reference:	SIP-ISUP/SS/CUG/	-	
SIP selection			
criteria:			
ISUP selection	PICS 5/7		
criteria:			
Test purpose:	Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "11"</cug>		
	Ensure that the <cugcommunicationindicator <cug="" a="" body="" call="" forward="" in="" ind="" is="" optional="" sent="" xml=""> <networkindicator> is mapped into the Network indentity and the XML <cug> <cug< th=""><th>dicator - CUG le IAM Closed</th><th>call indicator ="11". The XML User Group interlock code</th></cug<></cug></networkindicator></cugcommunicationindicator>	dicator - CUG le IAM Closed	call indicator ="11". The XML User Group interlock code
	IAM Closed User Group interlock code Binar		in your is mapped into the
SIP Parameter	INVITE:		
values:	<cug></cug>		
	<networkindicator>[PIXIT]<th></th><th></th></networkindicator>		
	<cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode>		
	<pre><cugcommunicationindicator>11</cugcommunicationindicator></pre>	mmunicationir	idicator>
ISUP Parameter	IAM:		
values:	Optional Forward Call Indicator CUG call indicator = "11"		
	Closed User Group interlock code		
	Binary code derived from INVITE XML be	ody < cugInter	lockBinaryCode>
	Network indentity derived from INVITE >	•	
Comments:	SIP	JT	ISUP
	INVITE →	→	IAM
	180 Ringing ←	+	ACM
		g tone	
	200 OK INVITE	+	ANM
	ACK →		
		rsation	DEI
	BYE ← 200 OK BYE →	←	REL
	200 OK BYE →	<u> </u>	RLC

TP516005	SIP reference: RFC 3261 [6]	ISUP reference: 7.4.1.1		
TSS reference:	SIP-ISUP/SS/CUG/			
SIP selection criteria:				
ISUP selection criteria:	NOT PICS 5/7			
Test purpose:	Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "10". The PSTN/ISDN network does not support CUG.</cug>			
	Ensure that the <cugcommunicationindicator> value "10" contained in the INVITE <cug> XML body is not sent in a optional forward call indicator - CUG call indicator = "10" when the PSTN/ISDN does not support CUG. If the optional forward call indicator have to be sent, the CUG call indicator is set to "00" no CUG call. No mapping of <networkindicator> and <cuginterlockbinarycode> into Closed User Group interlock code.</cuginterlockbinarycode></networkindicator></cug></cugcommunicationindicator>			
SIP Parameter	INVITE:	·		
values:	<pre><cug> <networkindicator>[PIXIT]</networkindicator> <cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode> <cugcommunicationindicator>10</cugcommunicationindicator> </cug></pre>			
ISUP Parameter	IAM:			
values:	Optional Forward Call Indicator CUG call indicator = "00" If optional forward call indicator have to be sent in case of an other indicator is not set to "0"			
Comments:	SIP	UT ISUP		
	INVITE ->	→ IAM		
	180 Ringing ←	← ACM		
	_	ng tone		
	200 OK INVITE ←	← ANM		
	ACK →			
		ersation		
	BYE •	← REL		
	200 OK BYE →	→ RLC		

TP516006	SID reference: D	EC 2264 [6]	1	ISUP reference:
17510000	SIP reference: R	FC 3201 [0]		
T00 (ES 283 027 [1], clause 7.4.1.1
TSS reference:	SIP-ISUP/SS/CUG/			
SIP selection				
criteria:				
ISUP selection	NOT PICS 5/7			
criteria:				
Test purpose:	Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "11". The PSTN/ISDN network does not support CUG.</cug>			
	Ensure that the <cugcommunicationindicator> value "11" contained in the INVITE <cug> XML body is sent in a optional forward call indicator - CUG call indicator = "11". The XML <cug> <networkindicator> is mapped into the IAM Closed User Group interlock code Network indentity and the XML <cug> <cuginterlockbinarycode> is mapped into the IAM Closed User Group interlock code Binary code.</cuginterlockbinarycode></cug></networkindicator></cug></cug></cugcommunicationindicator>			
SIP Parameter	INVITE:			
values:	<cug></cug>			
	<networkindicator>[PIXIT]</networkindicator>			
	<cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode>			
	< cugCommunicationI	ndicator>11 </th <th>cugCommunic</th> <th>ationIndicator></th>	cugCommunic	ationIndicator>
ISUP Parameter				
values:				
Comments:	SIP		SUT	ISUP
	INVITE	→		
	403 Forbidden	←		
	ACK	→		

6.3.2 Interworking from ISUP to SIP (Outgoing Call)

6.3.2.1 Calling Line Identification (CLI)

TP601001	SIP refe	rence: RFC 3261 [6]		ISUP reference:
			ES 28	3 027 [1], clause 7.2.3.1.2.6
TSS reference:	ISUP-SIP/SS/CI	_l/		
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:		number and no additional ca		
				e whereby Calling Party Number
	parameter and	the Generic Number are not of	containeu.	
	• Sands	an INVITE message without t	ha "P-Assar	ted-Identity header field" a
				ous.invalid and without a Privacy
	Heade		o earioriyiin	odoivana ana wiinodi a i iivaoy
SIP Parameter	INVITE: From <	unavailable@anonymous>		
values:		•		
ISUP Parameter	IAM: no Callir	ng party number		
values:	no Gene	eric Number: "Additional callin	g party num	ber"
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conve	ersation	
	REL	→	→	BYE
	RLC	←	←	200 OK BYE

TP601002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6		
TSS reference:	ISUP-SIP/SS/CLI/			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	No calling party number and additional calling party number received			
	Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter is not contained and the Generic Number is contained whereby the address presentation restriction parameter is set to "presentation allowed" and the Nature of Address Indicator is set to NoAS_VALUE: • Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" and no Privacy Header field.			
SIP Parameter values:	P-Asserted-Identity header field: not include: From header field: Tel or SIP URI: Addr_SPEC_ID Derived Signals (AcgPN)	d: from Generic Number parameter Address		
	Privacy header: is not included			
ISUP Parameter values:	IAM: Generic Number: "additional calling party number" Nature of Address Indicator: NoAS VALUE APRI "presentation allowed"			
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	ACM ←	← 180 Ringing		
	Ringing tone			
	ANM ←	← 200 OK INVITE		
		→ ACK		
	Conve	ersation		
	REL →	→ BYE		
	RLC ←	★ 200 OK BYE		

TP601003	SIP refere	nce: RFC 3261 [6]	ES 28	ISUP reference: 3 027 [1], clause 7.2.3.1.2.6
TSS reference:	ISUP-SIP/SS/CLI/			
SIP selection criteria:				
ISUP selection				
criteria:				
Test purpose: SIP Parameter values:	received Ensure that when parameter is not presentation restr Address Indicator • Sends an header field INVITE: From < P-Asset	the SUT has received an I. contained and the Generic riction parameter is set to "priss set to NoAS_VALUE: INVITE message without the set to unavailable@anonymous>erted-Identity header field: not the set to unavailable and the set to unavailable anonymous>	AM messag Number is coresentation he P-Assert nymous.inva	e whereby Calling Party Number contained whereby the address restricted" and the Nature of ed-Identity header field, a From alid and no Privacy Header field.
		y header: is not included		
ISUP Parameter values:		umber: "additional calling pa LUE APRI "presentation res		", Nature of Address Indicator:
	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
		Ringing tone		
Comments:	ANM	←	←	200 OK INVITE
			→	ACK
		Conve	rsation	
	REL	→	→	BYE
	RLC	←	←	200 OK BYE

Table 39

	Values for test purpose TP601003				
	ISUP Parameter values:	SIP Parameter values:			
VA_01	IAM NoAS_VALUE: "national (significant) number"(NDC+SN)	INVITE FHf_Addr_SPEC_ID: CC (of the country where the MGCF is located) is added to the Generic Number Address Signals and then mapped to user portion of URI scheme			
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.			

TP601004	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6		
TSS reference:	ISUP-SIP/SS/CLI/	ES 203 027 [1], clause 7.2.3.1.2.0		
SIP selection	ISUF-SIF/SS/CLI/			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Calling party number presentation allowed and no additional calling party number received			
	Ensure that when the SUT has received an Lacontained whereby the Nature of Address Inc. set to "presentation allowed" and the Generic	dicator is set to NoAS_VALUE the APRI is		
	"Tel or SIP URI" is set to PAIh_Addr_	or SIP URI" is set to FHf_Addr_SPEC_ID		
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)			
	From header field: Tel or SIP URI: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)			
	Privacy header: is not included or if included, "id" is not included			
ISUP Parameter values:	IAM: Calling party number APRI "presentat no Generic Number: "Additional callin	ion allowed"		
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	ACM ←	← 180 Ringing		
	Ringing tone			
	ANM ←	← 200 OK INVITE		
		→ ACK		
	Conve	rsation		
	REL →	→ BYE		
	RLC ←	★ 200 OK BYE		

Table 40

	Values for test purpose TP601004					
	ISUP Parameter values:	SIP Parameter values:				
VA_01	IAM	INVITE				
	NoAS_VALUE: "national (significant) number"(NDC+SN)	PAIh_Addr_SPEC_ID = FHf_Addr_SPEC_ID: Add CC (of the country where the MGCF is located) to CgPN Signals then map to user portion of URI scheme used				
VA_02	IAM NoAS_VALUE: "international number" ("+"CC+NDC+SN)	INVITE PAIh_Addr_SPEC_ID= FHf_Addr_SPEC_ID: the complete to CgPN Signals is mapped to the user portion of URI scheme.				

TP601005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6			
TSS reference:	ISUP-SIP/SS/CLI/	200 027 [1], clause 7.2.0.1.2.0			
SIP selection	1301 -311 /33/GEI/				
criteria:					
ISUP selection					
criteria:					
Test purpose:	Calling party number presentation restricted and no additional calling party number received Ensure that when the SUT has received an IAM message, the Calling Party Number is contained whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to "presentation restricted" and the Generic Number is not contained: • Sends an INVITE message with the "P-Asserted-Identity header field" where the "Tel or SIP URI" is set to PAIh_Addr_SPEC_ID, a "From header field" set to				
	unavailable@anonymous.invalid and wit	th Privacy Header field value "id".			
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) From header field: Tel or SIP URI Tel or SIP URI: unavailable@anonymous.invalid				
	Privacy header: "id".				
ISUP Parameter values:	IAM: Calling party number APRI "presentation no Generic Number: "Additional calling party number: "Additiona				
Comments:	ISUP/BICC SUT	SIP			
	IAM →	→ INVITE			
	ACM ←	← 180 Ringing			
	Ringing tone				
	ANM ←	← 200 OK INVITE			
	_	→ ACK			
		ersation			
	REL →	→ BYE			
	RLC ←	← 200 OK BYE			

TP601006	SIP reference: RF0	C 3261 [6]	ES 28	ISUP reference: 3 027 [1], clause 7.2.3.1.2.6	
TSS reference:	ISUP-SIP/SS/CLI/				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	Calling party number presentation restricted by the network and no additional calling party number received Ensure that when the SUT has received an IAM message, the Calling Party Number is				
				t to NoAS_VALUE the APRI is eneric Number is not contained:	
	"Tel or SIP URI" is		_SPEC_ID,	-Identity header field" where the a "From header field" set to er field value "id".	
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)				
	From header field: Tel or SIP URI Tel or SIP URI: unavailable@hostportion				
ISUP Parameter	Privacy header: "id". IAM: Calling party numb	er APRI "nresentat	ion restricte	d by the network"	
values:	no Generic Numbe				
Comments:	ISUP/BICC SUT SIP				
	IAM	→	→	INVITE	
	ACM ← 180 Ringing				
	_	ging tone			
	ANM	←	←	200 OK INVITE	
		_	→	ACK	
		_	rsation		
	REL)	→	BYE	
	RLC			200 OK BYE	

Table 41

	Values for test purpose TP601006					
	ISUP Parameter values:	SIP Parameter values:				
VA_01	NoAS_VALUE: "national (significant) number"(NDC+SN)	INVITE PAIh_Addr_SPEC_ID = FHf_Addr_SPEC_ID: CC (of the country where the MGCF is located) is added to the CgPN Signals and then mapped to user portion of URI scheme used				
VA_02	IAM NoAS_VALUE: "international number" ("+"CC+NDC+SN)	INVITE PAIh_Addr_SPEC_ID= FHf_Addr_SPEC_ID: the complete to CgPN Signals is mapped to the user portion of URI scheme.				

TP601007	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6			
TSS reference:	ISUP-SIP/SS/CLI/	, J,			
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	Calling party number presentation allowed and an additional calling party number received				
	Ensure that when the SUT has received an I contained whereby the Nature of Address Inc set to "presentation allowed" and the Generic	dicator is set to NoAS_VALUE the APRI is			
	Sends an INVITE message with the "P-Asserted-Identity header field", where the "Tel or SIP URI" is set to PAIh_Addr_SPEC_ID "From header field" where the "Tel or SIP URI" is set to FH_Addr_SPEC_ID and without "Privacy Header field" or "id" is not included.				
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)				
	From header field: Tel or SIP URI Tel or SIP URI: FH_Addr_SPEC_ID (Derived from Generic Number parameter Address Signals (AcgPN))				
	Privacy header: is not included or if included, "id" is not included.				
ISUP Parameter hbvalues:	IAM: Calling party number APRI "presentation allowed" Generic Number: "additional calling party number" Nature of Address Indicator: CP NoAS VALUE				
Comments:	ISUP/BICC SUT	SIP			
	IAM →	→ INVITE			
	ACM ←	← 180 Ringing			
	Ringing tone				
	ANM ←	€ 200 OK INVITE			
	0	→ ACK			
		rsation → BYE			
	REL → RLC ←	→ BYE ← 200 OK BYE			
	NLC T	▼ ZUU UN DIE			

TP601008	SIP reference: RFC 3261 [6]		ISUP reference:		
		ES 2	83 027 [1], clause 7.2.3.1.2.6		
TSS reference:	ISUP-SIP/SS/CLI/				
SIP selection					
criteria:					
ISUP selection					
criteria:					
Test purpose:	Calling party number presentation restricted and an additional calling party number received				
	Ensure that when the SUT has received an Lacontained whereby the Nature of Address Inc. set to presentation restricted and the General	dicator is se	et to NoAS_VALUE the APRI is		
	Sends an INVITE message with the				
	"P-Asserted-Identity header field, where the				
	"From header field" where the "addr-spec" i	s set to FH	_Addr_SPEC_ID and with		
010.0	"Privacy Header field =id".				
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)				
	From header field: addr-spec				
	Tel or SIP URI: FH_Addr_SPEC_ID (Derived from Generic Number				
	parameter Address Signals (AcgPN))				
	Privacy header: "id"				
ISUP Parameter	IAM: Calling party number APRI "presentat				
values:	Generic Number: "additional calling page 19				
	CP_NoAS_VALUE APRI: presentation restricted				
Comments:	ISUP/BICC SUT	_	SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	Ringing tone	←	200 OK INVITE		
	VIAIAI	→	ACK		
	Conve	rsation	, ion		
	REL →	→	BYE		
	RLC ←	+	200 OK BYE		

Table 42

	Values for test purpose TP601007; TP601008				
Test	ISUP Parameter values:	SIP Parameter values:			
purposes					
VA_01	IAM NoAS_VALUE: "national (significant) number"(NDC+SN)	INVITE FHf_Addr_SPEC_ID: Add CC (of the country where the MGCF is located) to Additional calling party number Signals then map to user portion of URI scheme used	PAIh_Addr_SPEC_ID: Add CC (of the country where the MGCF is located) to Calling party number Signals then map to user portion of URI scheme used		
VA_02	IAM NoAS_VALUE: "international number" ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete Additional calling party number Address Signals is mapped to the user portion of URI scheme.	PAIh_Addr_SPEC_ID: the complete Calling party number Address Signals is mapped to the user portion of URI scheme used.		

6.3.2.2 Call Hold (HOLD)

TP602001	SIP reference	: RFC 3261 [6]		ISUP reference:
				7.4.10/[14]
TSS reference:	ISUP-SIP/SS/HOLD/			
SIP selection	PICS 8/4			
criteria:				
ISUP selection	PICS 5/22			
criteria:				
Test purpose:	Each party can noid a	nd retrieve the remote p	arty in the c	confirmed state
				time after the call is answered an retrieve the call previously put
	on hold.		, ,	, ,,
		ald be able to put the other		
		ald be able to retrieve the ld be able to put the othe		
		ld be able to retrieve the		
SIP Parameter	SDP: a=sendonly (put		outer party	
values:		omitted (retrieve the call	l)	
	o= <version< th=""><th></th><th>•</th><th></th></version<>		•	
ISUP Parameter		tion: remote hold Event		
values:				PROGRESS (retrieve the call)
Comments:	ISUP/BICC	MGC	=	SIP
	IAM	→	→	INVITE
	ACM	←	(180 Ringing
	ANM	←	(200 OK INVITE
	CPG(hold)	→	→	INVITE(sendonly)
			←	200 OK INVITE(recvonly)
	CPG(retrieve)	→	→	INVITE(sendrecv)
			(200 OK INVITE(sendrecv)
	CPG(hold)	←	(INVITE(sendonly)
			→	200 OK INVITE(recvonly)
	CPG(retrieve)	←	←	INVITE(sendrecv)
			→	200 OK INVITE(sendrecv)

TP602002	SIP reference: RFC 3261 [6]	ISUP reference: 7.4.10/[14]			
TSS reference:	ISUP-SIP/SS/HOLD/				
SIP selection criteria:	PICS 8/4 AND PICS 8/1				
ISUP selection criteria:	PICS 5/22				
Test purpose:	The calling party can hold and retrieve the ren	note party in the early dialogue			
	Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold. The calling party should be able to retrieve the other party.				
SIP Parameter	SDP: a=sendonly (put on hold)	, ,			
values:	a=sendrecv or omitted (retrieve the call) o= <version incremented=""></version>				
ISUP Parameter	CPG: Generic notification: remote hold Event	ndicator PROGRESS (put on hold)			
values:	Generic notification: remote retrieval eve	nt indicator PROGRESS (retrieve the call)			
Comments:	ISUP/BICC MGC	F SIP			
	IAM →	→ INVITE			
	ACM ←	← 180 Ringing			
	CPG(hold) →	→ UPDATE(sendonly)← 200 OK UPDATE(recevonly)			
	CPG(retrieve) →	→ UPDATE(sendrecv)← 200 OK UPDATE(sendrecv)			

TP602003	SIP reference: RFC 3261 [6]		ISUP reference: 7.4.10/[14]		
TSS reference:	ISUP-SIP/SS/HOLD/				
SIP selection criteria:	PICS 8/4				
ISUP selection criteria:	PICS 5/22				
Test purpose:	HOLD indication in SDP in an UPDATE received Ensure that a party can put the other party on hold after the calling user has provided all of the information necessary for processing the call. Ensure that the party can retrieve the				
	call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party				
SIP Parameter	SDP: a=sendonly (put on hold)				
values:	a=sendrecv or omitted (retrieve the ca	II)			
ISUP Parameter	ACM: called party status: no indication				
values:	CPG: Generic notification: remote hold Event Generic notification: remote retrieval ev				
Comments:	ISUP/BICC MG0	F	SIP		
	IAM →	→	INVITE		
	ACM ←	←	180 Ringing		
	ANM ←	←	200 OK INVITE		
	UPDATE(sendonly) 200 OK UPDATE(recevonly)				
	CPG(retrieve)	←	UPDATE(sendrecv) 200 OK UPDATE(sendrecv)		

TP602004	SIP reference: RF	FC 3261 [6]		ISUP reference: 7.4.10/[14]	
TSS reference:	ISUP-SIP/SS/HOLD/				
SIP selection	PICS 8/4 AND PICS 8/3				
criteria:					
ISUP selection	PICS 5/22				
criteria:					
Test purpose:	The SUT uses the UPDA	TE method to indica	te HOLD in the	e SDP	
	Ensure that a party can po	ut the other party on	hold in the ale	erting state. Ensure that the	
	party can retrieve the call	previously put on ho	old.		
	The calling party should b	e able to put the oth	er party on ho	old	
	The calling party should b	e able to retrieve the	e other party		
SIP Parameter	SDP: a=sendonly (put on				
values:	a=sendrecv or omitt	ed (retrieve the call)			
	o= <version incre<="" th=""><th>mented></th><th></th><th></th></version>	mented>			
ISUP Parameter	CPG: Generic notification				
values:				PROGRESS (retrieve the call)	
Comments:	ISUP/BICC	MGC	F	SIP	
	IAM	→	→	INVITE	
	ACM	←		180 Ringing	
	ANM	←	←	200 OK INVITE	
	CPG(hold) → UPDATE(sendonly)				
			←	200 OK UPDATE(recevonly)	
	CPG(retrieve)	→	→	UPDATE(sendrecv)	
	, ,			200 OK UPDATE(sendrecv)	

TP602005	SIP reference: RFC	3261 [6]		ISUP reference: 7.4.10/[14]	
TSS reference:	ISUP-SIP/SS/HOLD/				
SIP selection criteria:	PICS 8/4				
ISUP selection	PICS 5/22				
criteria:					
Test purpose:	Each party can hold and retrieve the remote party in the confirmed state				
				time after the call is answered an retrieve the call previously put	
	The calling party should be able to put the other party on hold The called party should be able to put the other party on hold The calling party should be able to retrieve the other party The called party should be able to retrieve the other party				
SIP Parameter	SDP: a=sendonly (put on ho				
values:	a=sendrecv or omitte o= <version increm<="" td=""><th></th><td>1)</td><td></td></version>		1)		
ISUP Parameter	CPG: Generic notification: re				
values:				PROGRESS (retrieve the call)	
Comments:	ISUP/BICC	MGC		SIP	
	IAM	→	→	INVITE	
	ACM	←	←	180 Ringing	
	ANM	←	+	200 OK INVITE	
	CPG(hold)	→	→	INVITE(sendonly)	
			←	200 OK INVITE(recvonly)	
	CPG(hold)	←	← →	INVITE(inactive) 200 OK INVITE(inactive)	
	CPG(retrieve)	→	→	INVITE(recvonly) 200 OK INVITE(sendonly)	
	CPG(retrieve)	←	←	INVITE(sendrecv) 200 OK INVITE(sendrecv)	

TP602006	SIP reference	e: RFC 3261 [6]		ISUP reference:
				7.4.10/[14]
TSS reference:	ISUP-SIP/SS/HOLD/			
SIP selection	PICS 8/4			
criteria:				
ISUP selection	PICS 5/22			
criteria:	 			
Test purpose:	Each party can hold a	and retrieve the remote p	arty in the c	confirmed state
	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold.			
	The calling party should be able to put the other party on hold The called party should be able to put the other party on hold The called party should be able to retrieve the other party The calling party should be able to retrieve the other party			
SIP Parameter	SDP: a=sendonly (pu			
values:	a=sendrecv or o	omitted (retrieve the call)		
	o= <version i<="" td=""><td></td><td></td><td></td></version>			
ISUP Parameter		ation: remote hold Event		
values:				PROGRESS (retrieve the call)
Comments:	ISUP/BICC	MGC		SIP
	IAM	→	→	INVITE
	ACM	←	(180 Ringing
	ANM	←	+	200 OK INVITE
	CPG(hold)	→	→	INVITE(sendonly)
			←	200 OK INVITE(recvonly)
	CPG(hold)	←	←	INVITE(inactive)
			→	200 OK INVITE(inactive)
	CPG(retrieve)	←	←	INVITE(recvonly)
			→	200 OK INVITE(sendonly)
	CPG(retrieve)	→	→	INVITE(sendrecv) 200 OK INVITE(sendrecv)

6.3.2.3 Terminal portability (TP)

Void.

6.3.2.4 Conference calling (CONF)

TP604001	SIP reference: RFC 3261 [6	i]	NO	SN reference:
			ES 283 02	?7 [1], clause 7.4.14
TSS reference:	ISUP-SIP/SS/CONF/			
SIP selection	PICS 8/2			
criteria:				
ISUP selection	PICS 5/10			
criteria:				
Test purpose:	Establish and disconnect a Conferen	nce		
	Ensure that the SUT does not stop to			
	streams if a CPG message Generic			
SIP Parameter	GEN_NOT_VALUE was received du	ie to the C	ONF supplemen	tary service.
values:				
ISUP Parameter	CPG: Generic notification = GEN N	IOT VALI	IE	
values:	Cr G. Generic notification = GEN_I	IOI_VAL	JL	
Comments:	ISUP/BICC		SUT	SIP
O THINION CO.	IAM	→	→	
	ACM	É	÷	
	Ringing to	_	•	100 Kinging
	ANM	+	4	200 OK INVITE
	AINW	•	÷	
		Conver	-	AGIC
	CPG(Conference established)	→	Sation	
	or o(contended established)	•		
	CPG(Conference disconnected)	→		
		Conver	sation	
	REL	→		BYF
	RLC	É	=	200 OK BYE

Table 43

	Values for test purpose TP604001, TP604002				
	CPG→				
	Generic notification				
	GEN_NOT_VALUE				
VA_01	Conference established				
VA_02	Conference disconnected				

TP604002	SIP reference: RFC 3261	[6]		NGN reference:
			ES 283	3 027 [1], clause 7.4.14
TSS reference:	ISUP-SIP/SS/CONF/			
SIP selection	PICS 8/2			
criteria:				
ISUP selection	PICS 5/10			
criteria:				
Test purpose:	Isolate and reattach a Conference			
		.,		
	Ensure that the SUT stop the temp			
	CPG message Generic notification			GEN_NOT_VALUE was
SIP Parameter	received due to the CONF supplem SDP: a= a LINE VA (see table 4			
values:	SDI . a= a_LINE_VA (see table 4	o, or a lifte	is offilited	
ISUP Parameter	CPG: Generic notification = Confe	rence estal	hlishad	
values:	CPG: Generic notification = GEN_			
Comments:	ISUP/BICC		SUT	SIP
	IAM	→ `		INVITE
	ACM	←	-	180 Ringing
	Ringing ton	e	_	100 14.119.119
	ANM	~	←	200 OK INVITE
	7			ACK
		Conve	rsation	-
	CPG(Conference established)	→		
	,			
	CPG(Isolated)	→	→	INVITE(sendonly)
	,		←	* * * * * * * * * * * * * * * * * * * *
			→	ACK
				-
	CPG(Reattached)	→	→	INVITE(sendrecv)
	,		←	*
			→	ACK
	CPG(Conference disconnected)	→		
	,	Conve	rsation	
	REL	→	→	BYE
	RLC	←	←	200 OK BYE

Table 44

Values for test purpose TP604001, TP604002					
	CPG→ INVITE/UPDATE→				
	Generic notification SDP attribute line				
	GEN_NOT_VALUE	a_LINE_VA			
VA_01 isolated		sendonly or inactive			
VA_02	VA_02 reattached sendrecv or recvonly or omitted				

TP604003	SIP reference: RFC 3261	[6]	ISUP reference: ITU-T Rec Q.1912.5 [32], annex B.14
			ITU-T Rec Q.734.1 [34], clause 1.7
TSS reference:	ISUP-SIP/SS/CONF/		
SIP selection			
criteria:	NOT DIOC 5/40		
ISUP selection criteria:	NOT PICS 5/10		
Test purpose:	Mapping of isolated and reattached	not sunna	orted
rest purpose.	Inapping of isolated and realization	ι ποι σαρρο	on ted
	Ensure that the MGCF can receive	in a CPG t	the Generic notifications "isolated" and
	"reattached", no mapping on the S	P side and	I the call is not disrupted.
			•
	No mapping, no disrupting the	SIP proced	ure.
SIP Parameter	No mapping		
values:			
ISUP Parameter values:	CPG: Generic notification = Confe		blished
values:	CPG: Generic notification = isolate CPG: Generic notification = reatta		
	CPG: Generic notification = Confe		onnected
Comments:	ISUP/BICC		SUT SIP
	IAM	→	→ INVITE
	ACM	←	← 180 Ringing
	Ringing ton	е	
	ANM	←	← 200 OK INVITE
			→ ACK
			ersation
	CPG(Conference established)	→	
	CPG(Isolated)	→	
	CPG(Reattached)	→	
	CPG(Conference disconnected)	→ Conve	ersation
	REL	→	→ BYE
	RLC	←	← 200 OK BYE

TP604004	SIP reference: RFC 3261	[6]		ISUP reference: c Q.1912.5 [32], annex B.14 c Q.734.1 [34], clause 1.7	
TSS reference:	ISUP-SIP/SS/CONF/				
SIP selection					
criteria:	NOT PICS 5/10				
ISUP selection criteria:	NOT PICS 5/10				
Test purpose:	No mapping of generic notifications no change the session state				
	Ensure that the MGCF can receive in a CPG the Generic notifications is "other party added" or "other party isolated" or "other party reattached" or "other party split" or " conference floating" or " other party disconnected" and there is no mapping on the SIP side and the call is not disrupted. No mapping, no disrupting the SIP procedure.				
SIP Parameter	No mapping				
values: ISUP Parameter	CPG: Generic notification = Confe	rongo onto	bliobod		
values:	CPG: Generic notification = Confe CPG: Generic notification = isolate		Diisnea		
values.	CPG: Generic notification = reatta				
	CPG: Generic notification = Confe	erence disc	onnected		
Comments:	ISUP/BICC IAM ACM Ringing ton	→	SUT → ←	SIP INVITE 180 Ringing	
	ANM	←	← → ersation	200 OK INVITE ACK	
	CPG(Conference established)	→	i Sation		
	CPG(other party added)	→			
	CPG(other party isolated)	→			
	CPG(other party reattached)	→			
	CPG(other party split)	→			
	CPG(other party disconnected)	→			
	CPG(Conference floating)	→			
	CPG(Conference disconnected)	→ Conve	rsation		
	REL RLC	→	→	BYE 200 OK BYE	

TP604005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.4.1.1.1		
TSS reference:	ISUP-SIP/SS/CONF/			
SIP selection criteria:	PICS 1/1			
ISUP selection criteria:				
Test purpose:	Conference notification information is a	mapped into "conference established"		
	element active is set to "true", the MG0 notification "conference established"			
SIP Parameter	NOTIFY 1: Event contains conferen	nce; Subscription-State contains active;		
values:	expires=xxxx			
	application/conference-i	nfo+xml:		
	<conference-info></conference-info>	1101 ()		
	entity=conference <conference-stat< th=""><th>e URI state="full" version="x"</th></conference-stat<>	e URI state="full" version="x"		
		e> 2 if present		
		if present		
	<users></users>			
	<user <="" entity="ISUPx" state="full" th="" uri=""></user>			
		entity=endpoint ISUPx URI		
		s>connected		
	<joinin< th=""><th>g-method>dialed-out<!-- joining-method--></th></joinin<>	g-method>dialed-out joining-method		
	<medi< th=""><th>a id="1"</th></medi<>	a id="1"		
		tatus>sendrecv		
		SIPx URI state="full"		
	·	entity=endpoint SIPx URI		
		s>connected		
		g-method>dialed-in joining-method a id="1"		
		a id= 1 tatus>sendrecv		
ISUP Parameter values:	CPG(Conference established)	adus/seriurecv\/status/		
Comments:	ISUP	MGCF SIP		
	IAM	→ INVITE		
	ACM	← 180 Ringing		
	ACM	← ← 200 OK INVITE		
	CPG(Conference established)	← NOTIFY 1→ 200 OK NOTIFY		
	REL	→ B YE		
	RLC	← 200 OK BYE		

TP604006	SIP reference: RFC 3261 [6]	ISUP refer ES 283 02	rence: ?7 [1], clause 7.4.1.1.1
TSS reference:	ISUP-SIP/SS/CONF/		
SIP selection	PICS 1/1		
criteria:			
ISUP selection criteria:			
Test purpose:	Conference notification information is	mapped into "confe	erence established"
	Upon the receipt of a conference info element status of endpoint-status-typ before and the Contact URI in the ele PSTN/ISDN participant, the MGCF statistication "other party added".	e is set to "connecte ment entity is not th	ed" and it was not set to "on-hold" ne address of the served
SIP Parameter values:	NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains conference application/conference <conference-info> entity=conference-sta <user-count <users=""> <user <endpoir="" <user-count="" <user-count<="" entity:="" td=""><td>-info+xml: ce URI state="full" vate> >y</td></user></user-count> if p =SIPx URI state="full" value in the entity=endpoint Sus>connected</conference-info>	-info+xml: ce URI state="full" vate> >y	version="x" present III" IPx URI atus> but joining-method>
ISUP Parameter	CPG(other party added)		
values:			
Comments:	ISUP IAM ACM ACM	MGCF → ←	SIP → INVITE ← 180 Ringing ← 200 OK INVITE
	CPG(Conference established)	←	NOTIFY 1→ 200 OK NOTIFY
	CPG(other party added)	←	NOTIFY 2→ 200 OK NOTIFY
	REL RLC	→ ←	→ BYE ← 200 OK BYE

TP604007	SIP reference: RFC 3261 [6]	ISUP refer ES 283 027	ence: 7 [1], clause 7.4.1.1.1
TSS reference:	ISUP-SIP/SS/CONF/		
SIP selection criteria:	PICS 1/1		
ISUP selection criteria:			
Test purpose:	Conference notification information is m	apped into "isolate	ed"
	Upon the receipt of a conference inform element status of endpoint-status-type is before and the Contact URI in the element participant, the MGCF shall send a CPG "isolated".	s set to "on-hold" a ent entity is the ad	and it was set to "connected" dress of the served PSTN/ISDN
SIP Parameter values:	<users> <user <u<="" <usee="" <user="" entity="IS" status:="" td=""><td>o+xml: URI state="full" ve> if pr UPx URI state="fu entity=endpoint ISI >on-hold -method>dialed-o</td><td>ersion="x" resent ull" UPx URI > ut<!-- joining-method--></td></user></users>	o+xml: URI state="full" ve> if pr UPx URI state="fu entity=endpoint ISI >on-hold -method>dialed-o	ersion="x" resent ull" UPx URI > ut joining-method
ISUP Parameter	CPG(isolated)		
values:	loup	11005	OID
Comments:	ISUP IAM ACM ACM	-	SIP → INVITE ← 180 Ringing ← 200 OK INVITE
	CPG(Conference established)	=	NOTIFY 1→ 200 OK NOTIFY
	CPG(isolated)	<u>-</u>	NOTIFY 2→ 200 OK NOTIFY
	REL -		→ BYE← 200 OK BYE

TP604008	SIP reference: RFC 3261 [6]	ISUP referes 283 02	rence: 27 [1], clause 7.4.1.1.1	
TSS reference:	ISUP-SIP/SS/CONF/		= =	
SIP selection criteria:	PICS 1/1			
ISUP selection criteria:				
Test purpose:	Conference notification information is Upon the receipt of a conference info element status of endpoint-status-typ before and the Contact URI in the ele PSTN/ISDN participant, the MGCF s notification "other party isolated".	ormation document vole is set to "on-hold" ement entity is not the	with the <endpoint-type> and the and it was set to "connected" ne address of the served</endpoint-type>	
SIP Parameter values:	NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains conference; Subscription-State contains active application/conference-info+xml: <conference-info> entity=conference URI state="full" version="x" <conference-state> <user-count>2</user-count> if present <users> <user <endpoint="" <status="" entity="endpoint" sipx="" state="full" uri="">on-hold <joining-method>dialed-out</joining-method> <media <="" id="1" td=""></media></user></users></conference-state></conference-info>			
ISUP Parameter values:	CPG(other party isolated)	status>sendrecv <th>idius/</th>	idius/	
Comments:	ISUP IAM ACM ACM CPG(Conference established) CPG(other party isolated) REL RLC	MGCF ← ← ←	SIP → INVITE ← 180 Ringing ← 200 OK INVITE ← NOTIFY 1 → 200 OK NOTIFY ← NOTIFY 2 → 200 OK NOTIFY → BYE ← 200 OK BYE	

SIP reference: RFC 3261 [6]	ISUP references 283 027 [1	ce:], clause 7.4.1.1.1	
ISUP-SIP/SS/CONF/			
PICS 1/1			
Upon the receipt of a conference infor element status of endpoint-status-type before and the Contact URI in the element status of endpoint in the element status of endpoint status of e	mation document with e is set to "connected" ment <i>entity</i> is the addre	the <endpoint-type> and the and it was set to "on-hold" ess of the served PSTN/ISDN</endpoint-type>	
NOTIFY 2: Event contains conference; Subscription-State contains active application/conference-info+xml: <conference-info> entity=conference URI state="full" version="x" <conference-state> <user-count>2</user-count> if present <users> <user <endpoint="" <status="" entity="endpoint" isupx="" state="full" uri="">on-hold <ipoining-method>dialed-out joining-method> <media <status="" id="1">sendrecv NOTIFY 3: Event contains conference; Subscription-State contains active application/conference-info+xml: <conference-info> entity=conference URI state="full" version="x" <conference-state> <user-count>2</user-count> if present <user> <user-count>2</user-count> if present <user> <user-count>2</user-count> ISUPx URI <user-count> if present</user-count></user> <user-count> if present</user-count></user> <user-count> if present</user-count></conference-state></conference-info></media></ipoining-method></user> <user-count> if present <user-< th=""></user-<></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></users></conference-state></conference-info>			
<\$	tatus>sendrecv <th>S></th>	S>	
CPG(reattached)			
		SIP	
		3 3	
ACM	+ +	200 OK INVITE	
CPG(Conference established)	+ +	NOTIFY 1	
	→	_	
CPG(isolated)	← ← →		
CPG(reattached)	← ← →	NOTIFY 3	
REL RLC	→ →	BYE 200 OK BYE	
	ISUP-SIP/SS/CONF/ PICS 1/1 Conference notification information is Upon the receipt of a conference infor element status of endpoint-status-type before and the Contact URI in the elemanticipant, the MGCF shall send a CF "reattached". NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains conference-info> entity=conference-conference-info> entity=conference-conference-info> entity=conference-conference-info> entity=conference-statuser-count> <users> <user entity="<conference-info"> entity=conference-conference-info> entity=conference-statuser-count> <users> <user entity="<conference-statuser-count"> <users> <user-count> <users> <user entity="<conference-statuser-count"> <users- <user-count=""> <user-count> <u< td=""><td> ISUP-SIP/SS/CONF/ PICS 1/1 </td></u<></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></user-count></users-></user></users></user-count></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users></user></users>	ISUP-SIP/SS/CONF/ PICS 1/1	

TP604010	SIP reference: RFC 3261 [6]	ISUP refe ES 283 0		e: , clause 7.4.1.1.1	
TSS reference:	ISUP-SIP/SS/CONF/				
SIP selection	PICS 1/1				
criteria:					
ISUP selection					
criteria:				· ·	
Test purpose:	Conference notification information is mapped into "reattached" Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "connected" and it was set to "on-hold" before and the Contact URI in the element entity is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "other party reattached".</endpoint-type>				
SIP Parameter	NOTIFY 1: see test case TP604006				
values:	NOTIFY 2: Event contains conference; Subscription-State contains active application/conference-info+xml: <conference-info> entity=conference URI state="full" version="x" <conference-state> <user-count>2</user-count> if present <users> <user <endpoint="" <status="" entity="endpoint" sipx="" state="full" uri="">on-hold <joining-method>dialed-out</joining-method> <media <status="" id="1">sendrecv NOTIFY 3: Event contains conference; Subscription-State contains active application/conference-info+xml: <conference-info> entity=conference URI state="full" version="x" <conference-state> <user-count>2</user-count> if present <user-count>2</user-count> IPx URI <user-conference-full" <endpoint="" <status="" entity="endpoint" sipx="" state="full" uri="">connected <joining-method>dialed-out</joining-method></user-conference-full"></conference-state></conference-info></media></user></users></conference-state></conference-info>				
	<media <status="" id="1">sendrecv</media>				
ISUP Parameter	CPG(other party reattached)				
values:					
Comments:	ISUP	MGCF		SIP	
	IAM	→	→	INVITE	
	ACM	←	←	180 Ringing	
	ACM	←	←	200 OK INVITE	
	CPG(Conference established)	←	← →	NOTIFY 1 200 OK NOTIFY	
	CPG(other party isolated)	←	← →	NOTIFY 2 200 OK NOTIFY	
	CPG(other party reattached)	←	←	NOTIFY 3 200 OK NOTIFY	
	REL RLC	→	→	BYE 200 OK BYE	

TP604011	SIP reference: RFC 3261 [6]	ISUP referen ES 283 027 [ce: 1], clause 7.4.1.1.1		
TSS reference:	ISUP-SIP/SS/CONF/				
SIP selection	PICS 1/1				
criteria:					
ISUP selection criteria:					
Test purpose:	Conference notification information is mapped into "other party disconnected"				
	Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "disconnected" and the element joining-method of joining-type is not set to "focus-owner, the MGCF shall send a CPG message to the CS side with a notification "other party disconnected".</endpoint-type>				
SIP Parameter	NOTIFY 1: see test case TP604006				
values:	NOTIFY 2: Event contains conference; Subscription-State contains active application/conference-info+xml: <conference-info></conference-info>				
	entity=conference URI state="full" version="x" <conference-state></conference-state>				
	<user-count>y</user-count> if present				
	<users> <user entity:<="" th=""><th>=SIPx URI state="full"</th><th></th></user></users>	=SIPx URI state="full"			
	<endpoint <status="" entity="endpoint" sipx="" uri="">connected <joining-method>dialed-out</joining-method> <media <="" id="1" td=""></media></endpoint>				
	<pre><status>sendrecv</status> NOTIFY 3: Event contains conference; Subscription-State contains active</pre>				
	application/conference-info+xml: <pre></pre>				
	entity=conference URI state="full" version="x"				
	<conference-state></conference-state>				
	<user-count>1</user-count> if present				
	<use s=""></use>				
	<user <endpoint="" <status="" entity="endpoint" sipx="" state="full" uri="">disconnected <joining-method>dialed-out</joining-method> <disconnection-method>booted<disconnection-method></disconnection-method> <media <status="" id="1">sendrecv</media></disconnection-method></user>				
ISUP Parameter	CPG(other party disconnected)				
values:					
Comments:	ISUP	MGCF	SIP		
	IAM	-	=		
	ACM	+ +	3 3		
	ACM	←	200 OK INVITE		
	CPG(Conference established)	+ •	NOTIFY 1		
	Ci G(Conference established)	+			
		_	200 01(11011111		
	CPG(other party added)	(NOTIFY 2		
		-	200 OK NOTIFY		
	CPG(other party disconnected)	(NOTIFY 3		
		-			
	REL	→ -	BYE		
	RLC	(

TP604012	SIP reference: RFC 3261 [6]	ISUP referen	
TSS reference:	ICLID CID/CC/CONE/	ES 283 027 [*], clause 7.4.1.1.1
SIP selection	ISUP-SIP/SS/CONF/ NOT PICS 1/1		
criteria:	NOT PICS 1/1		
ISUP selection			
criteria:			
Test purpose:	Conference notification information is no Upon the receipt of a conference information is not mapped to the PSTN s	ation document the o	
SIP Parameter values:	NOTIFY 1: Event contains conference expires=xxxx NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains conference application/conference-info < conference-info > entity=conference < conference-state > cuser-count>y< < user < entity=Sll < endpoint e < status> < joining-cmedia i	e; Subscription-State c+xml: JRI state="full" vers /user-count> if pres Px URI state="full" ntity=endpoint SIPx connecteddialed-out<	e contains active; e contains active ion="x" ent URI > x/ joining-method>
ISUP Parameter values:	CPG(other party added)		
Comments:	REL RLC	← → →	180 Ringing 200 OK INVITE NOTIFY 1 200 OK NOTIFY NOTIFY 2 200 OK NOTIFY BYE

TP604013	SIP reference: RFC 3261 [6]	ISUP reference:	
TSS reference:	ISUP-SIP/SS/CONF/	•	
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	The referring of MGCF is not possible w	vhen a call is established	
	Ensure that a REFER request received by the MGCF is not successful. The request is rjected with . 403 Forbidden. The CS -site is not affected.		
SIP Parameter values:	REFER: Request URI contained the contained the Contained the URI of Referred By contained SIR or the Contained Referred	ISUPx, method=invite	
ISUP Parameter	Referred-By contains SIP or t CPG(Conference established)	tel URI di SIPX	
values:	or o(comerence established)		
Comments:	ISUP	MGCF SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	ACM ←	← 200 OK INVITE	
		← REFER	
		→ 403 Forbidden	

6.3.2.5 Three Party service (3PTY)

TP605001	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [1], clause 7.4.15
TSS reference:	ISUP-SIP/SS/3PTY/		
SIP selection	PICS 8/2		
criteria:			
ISUP selection	PICS 5/5 AND PICS 5/18		
criteria:			
Test purpose:	The media stream is resumed if a 3	PTY is esta	tablished
		-INVITE co	edia stream put on hold while the GPG (hold) containing an a-line in the SDP is set to ablished) was received
SIP Parameter	SDP: a= a_LINE_VA (see table 45	5) or a line	is omitted
values:	·		
ISUP Parameter	CPG: Generic notification = remote	e hold	
values:	CPG: Generic notification = sendr		
Comments:	ISUP/BICC	S	SUT SIP
	IAM	→	→ INVITE
	ACM	←	 180 Ringing
	Ringing tone	Э	
	ANM	←	◆ 200 OK INVITE
			→ ACK
		Conve	ersation
	CPG(hold)	→	→ INVITE(sendonly)
			← 200 OK INVITE(recvonly)→ ACK
	CPG(Conference established)	→	→ INVITE(sendrecv)← 200 OK INVITE(sendrecv)→ ACK
	CPG(Conference disconnected)	→ Convei	ersation
	REL	→	→ BYE
	RLC	←	← 200 OK BYE

TP605002	SIP reference: RFC 3261	[6]	NGN reference:
			ES 283 027 [1], clause 7.4.15
TSS reference:	ISUP-SIP/SS/3PTY /		
SIP selection	PICS 8/1		
criteria:			
ISUP selection	PICS 5/5 AND PICS 5/18		
criteria:			
Test purpose:	Establish and disconnect a 3PTY s	ession. SD	P conveyed in an UPDATE request
		UPDATE c	dia stream put on hold while the GPG (hold) ontaining an a-line in the SDP is set to blished) was received
SIP Parameter values:	SDP: a= a_LINE_VA (see table 4	5) or a line	is omitted
ISUP Parameter	CPG: Generic notification = remot	e hold	
values:	CPG: Generic notification = GEN _	NOT_VAL	UE
Comments:	ISUP/BICC	5	SUT SIP
	IAM	→	→ INVITE
	ACM	(← 180 Ringing
	Ringing ton	е	
	ANM	←	← 200 OK INVITE→ ACK
		Conve	rsation
	CPG(hold)	→	→ UPDATE(sendonly)← 200 OK UPDATE(recvonly)
	CPG(Conference established)	→	→ UPDATE(sendrecv)← 200 OK UPDATE(sendrecv)
	CPG(Conference disconnected)		rsation
	REL	→	→ BYE
	RLC	+	★ 200 OK BYE

Table 45

	Values for test purpose TP605001, TP605002		
	CPG→ INVITE/UPDATE→		
	Generic notification SDP attribute line		
GEN_NOT_VALUE a_LINE_VA			
VA_01	Conference established	sendrecv, or recvonly	
VA_02	Conference disconnected	sendrecv or recvonly	

TP605003	SIP reference: RFC 3261 [6	6]	ISUP reference: ES 283 027 [1], clause 7.4.13
TCC voferonce:	IOUR OID/OO/ODTV/		ITU-T Rec Q.734.2 [35], clause 2.7
TSS reference:	ISUP-SIP/SS/3PTY/		
SIP selection criteria:			
ISUP selection criteria:	NOT PICS 5/18		
Test purpose:		CPG mess cator with	
SIP Parameter values:	No mapping		
ISUP Parameter	CPG: Generic notification = remote	hold	
values:	CPG: Generic notification = Confer CPG: Generic notification = Confer		
Comments:	ISUP/BICC		SUT SIP
	IAM	→	→ INVITE
	ACM	(← 180 Ringing
	Ringing tone)	5 5
	ANM	←	← 200 OK INVITE → ACK
	CPG(hold)	Conver	
		→	
	CPG(Conference disconnected)	→ Conver	rsation
	REL	→	→ BYE
	RLC	(← 200 OK BYE

6.3.2.6 Connected line identification (COL)

TP606001	SIP reference: RF0	C 3261 [6]	ES 28	3 027 [1], clause 7.4.2
TSS reference:	ISUP-SIP/SS/COL /			
SIP selection	NOT PICS 5/3			
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM with OFCI "connected	line request" receiv	∕ed, no mapping	1
				ward call indicator, connected
	line requested, continue with	thout disrupting the	SIP or ISUP sig	gnalling procedure.
SIP Parameter	No mapping	No mapping		
values:				
ISUP Parameter	IAM: Optional Forward call	indicator "Connect	ed line request"	
values:				
Comments:				
	ISUP	5	SUT	SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
	REL	→	→	BYE
	RLC	←	←	200 OK BYE

TP606002	SIP reference: RFC 3261 [6]	ES 28	3 027 [1], clause 7.4.2
TSS reference:	ISUP-SIP/SS/COL /		
SIP selection	PICS 5/3		
criteria:			
ISUP selection			
criteria:			
Test purpose:	IAM with oFCi "connected line request" received, INVITE is sent contains the "from-change" tag in the Supported header		
	Ensure that the SUT if the IAM is received		
	line requested, the "from-change" tag is in	cluded in the Suppor	rted header in the sent INVITE.
SIP Parameter	INVITE: Supported: from-change		
values:			
ISUP Parameter values:	IAM: Optional Forward call indicator "Con	nected line request"	
	ISUP	SUT	SIP
	IAM →	→	INVITE
	ACM ←	←	180 Ringing
	ANM ←	←	200 OK INVITE
		→	ACK
	C	onversation	
	REL →	→	BYE
	RLC ←	+	200 OK BYE

TP606003	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.4.2	
TSS reference:	ISUP-SIP/SS/COL /		
SIP selection criteria:	PICS 5/3		
ISUP selection	NOT PICS 1/5		
criteria:	INOT PICS 1/5		
Test purpose:	The P-Asserted-Identity is mapped into the connected number "national (significant) number"		
	been requested by the calling party by parsin	of an IAM message where the COLP service has g the "Optional Forward Call Indicators" field and or" is set to "requested", on receipt of a 1XX or A with	
	NDC+ SN has been received and the	ining a URI with an identity in the format "+" CC+ Privacy header field was received and the priv-	
	in the ANM or CON is included the Connecte If CC encoded in the URI is equal to the CC next BICC/ISUP node is located in the same	of the country where MGCF is located AND the	
	Address presentation restricted parameter = presentation allowed Nature of address indicator = National (significant) number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network Provided Address signals in the format: NDC+SN		
	Generic number parameter not present		
SIP Parameter	1XX or 2XX response: P-Asserted-Identity he	eader field Tel LIRL containing an LIRL in the	
values:	format "+"CC+NDC+SN	addi noid for one domaining an one in the	
ISUP Parameter	IAM: Optional Forward Call Indicators, Conne	ected Line Identity Request indicator" =	
values:	"requested"	, .	
	ANM;		
	Connected number parameter		
	Address presentation restricted paramet	er = '00'B	
	Nature of address indicator = '0000011'B		
	Numbering plan indicator = '001'B		
	Screening indicator = '11'B		
	Address signals = PIXIT		
		MGCF SIP	
	IAM →	→ INVITE ★ SIP_MESSAGE_VA	
	CASE A		
	ACM ←		
	ANM ←		
	CASE B		
	CON +		
		versation versation	
	REL →	→ BYE	
	RLC +	← 200 OK BYE	

TP606004	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.4.2		
TSS reference:	ISUP-SIP/SS/COL /			
SIP selection	PICS 5/3			
criteria:	1 100 3/3			
ISUP selection	PICS 1/5			
criteria:				
Test purpose:	The P-Asserted-Identity is mapped into the co	onnected number "international number"		
	been requested by the calling party by parsing the "Connected Line Identity Request indicate 2XX message defined as SIP_MESSAGE_VA the P-Asserted-Identity header field contain NDC+ SN has been received and the	Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the "Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with the P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the		
	value is set to "none"	Privacy header field was received and the priv-		
	in the ANM or CON is included the Connecte If CC encoded in the URI is not equal to the 0 the next BICC/ISUP node is located in the sar	CC of the country where MGCF is located AND		
	Address presentation restricted parameter = Presentation allowed Nature of address indicator = International number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network Provided Address signals in the format: CC+NDC+SN			
	Generic number parameter not present			
SIP Parameter values:	1XX or 2XX response: P-Asserted-Identity he format "+"CC+NDC+SN	ader field Tel URL containing an URI in the		
ISUP Parameter	IAM: Optional Forward Call Indicators, Conne	cted Line Identity Request indicator" =		
values:	"requested"	, ,		
	ANM;			
	Connected number parameter			
	Address presentation restricted parameter			
	Nature of address indicator = '0000011'B			
	Numbering plan indicator = '001'B			
	Screening indicator = '11'B			
	Address signals = PIXIT	CCE SID		
	ISUP M	GCF SIP → INVITE		
	IOIVI 7	SIP_MESSAGE_VA		
	CASE A	5E00/(OE_V/(
	ACM			
	ANM ←			
	CASE B			
	CON			
		ersation		
	REL →	→ BYE		
	RLC +	€ 200 OK BYE		

TP606005	SIP reference: RFC 3261 [6]		SUP reference:
		ES 283	3 027 [1], clause 7.4.2
TSS reference:	ISUP-SIP/SS/COL /		
SIP selection	PICS 5/3		
criteria:			
ISUP selection	PICS 1/5		
criteria:			
Test purpose:	P-Asserted-Identity not received, a connected number "address not available" is sent		
	Ensure that the SUT in Idle state, on receipt of been requested by the calling party by parsing the "Connected Line Identity Request indicato 2XX message defined as SIP_MESSAGE_VA	the "Optional For " is set to "reque	orward Call Indicators" field and
	no P-Asserted-Identity header field		
	In the ANM or CON is included the Connected	d number Paran	neter.
	Address presentation restricted parameter Screening indicator = Network Provided Address signals omitted	= Address not a	available
	Generic number parameter not present		
SIP Parameter	1XX or 2XX response: P-Asserted-Identity hea	ader field is not p	resent
values:	That or End the species in the second and the second secon		
ISUP Parameter	IAM: Optional Forward Call Indicators, Connec	ted Line Identity	Request indicator" =
values:	"requested"	·	·
	ANM or CON		
	Connected number parameter		
	Address presentation restricted paramete	r = '10'B	
	Nature of address indicator = '0000000'B		
	Numbering plan indicator = '000'B		
	Screening indicator = '11'B		
	Address signals = not presented		
	ISUP M	GCF	SIP
	IAM →	→	INVITE
		←	SIP_MESSAGE_VA
	CASE A		
	ACM ←		
	ANM ←		
	CASE B		
	CON +		
		ersation	
	REL →	→	BYE
	RLC ←	<u> </u>	200 OK BYE

	Values for tests purposes TP101003 to TP101005
VA_01	180 Ringing
VA_02	183 Session progress
VA 03	200 OK

TP606006	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.4.2	
TSS reference:	ISUP-SIP/SS/COL /		
SIP selection criteria:	PICS 5/3		
ISUP selection criteria:	NOT PICS 1/5		
Test purpose:	Interworking of P-Asserted-Identity. Convert	the user portion into the national format	
	been requested by the calling party by parsin	of an IAM message where the COLP service has ig the "Optional Forward Call Indicators" field and or" is set to "requested", on receipt of a 1XX or A with	
	the P-Asserted-Identity header field conta NDC+ SN has been received and the a Privacy header field was received and the	nining a URI with an identity in the format "+" CC+ he priv-value is set to PRIV_VALUE	
	in the ANM or CON is included the Connecte If CC encoded in the URI is equal to the CC next BICC/ISUP node is located in the same	of the country where MGCF is located AND the	
	Address presentation restricted parameter = Presentation restricted Nature of address indicator = National (significant) number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network Provided Address signals in the format: NDC+SN		
	Generic number parameter not present		
SIP Parameter values:	1XX or 2XX response: P-Asserted-Identity he format "+"CC+NDC+SN	eader field Tel URL containing an URI in the	
ISUP Parameter	IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" =		
values:	"requested"		
	ANM or CON		
	Connected number parameter		
	Address presentation restricted parameter		
	Nature of address indicator = "0000011"I	В	
	Numbering plan indicator = "001"B		
	Screening indicator = "11"B		
	Address signals = PIXIT		
		MGCF SIP	
	IAM →	→ INVITE← SIP_MESSAGE_VA	
	CASE A		
	ACM ←		
	ANM ←		
	CASE B		
	CON +		
		versation	
	REL -	→ BYE	
	RLC ←	← 200 OK BYE	

TP606007	SIP reference: RFC 3261 [6]	ISUP reference:		
		ES 283 027 [1], clause 7.4.2		
TSS reference:	ISUP-SIP/SS/COL /			
SIP selection criteria:	PICS 5/3			
ISUP selection	PICS 1/5			
criteria:				
Test purpose:	Interworking of P-Asserted-Identity. Convert to	he user portion into the international format		
	Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the "Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with			
	the P-Asserted-Identity header field contain NDC+ SN has been received and the a Privacy header field was received and the	ning a URI with an identity in the format "+" CC+ ne priv-value is set to PRIV_VALUE		
	in the ANM or CON is included the Connecte If CC encoded in the URI is not equal to the 0 the next BICC/ISUP node is located in the sar	CC of the country where MGCF is located AND		
	Address presentation restricted parameter = Presentation restricted Nature of address indicator = International number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network Provided Address signals in the format: CC+NDC+SN			
	Generic number parameter not present			
SIP Parameter	1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the			
values:	format "+"CC+NDC+SN	ado:o.d . o. o ooag a o u.o		
ISUP Parameter values:	IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested"			
values.	requested			
	ANM or CON			
	Connected number parameter			
	Address presentation restricted parameter	er = '01'B		
	Nature of address indicator = '0000011'B			
	Numbering plan indicator = '001'B			
	Screening indicator = '11'B			
	Address signals = PIXIT			
	ŭ	GCF SIP		
	IAM →	→ INVITE← SIP_MESSAGE_VA		
	CASE A			
	ACM ←			
	ANM ←			
	CASE B			
	CON ←			
	Conv	ersation		
	REL →	→ BYE		
	RLC +	← 200 OK BYE		

TP606008	SIP reference: RFC 3261 [6]	ES 283 02	ISUP reference: 27 [1], clauses 7.4.2 and 7.5.2
TSS reference:	ISUP-SIP/SS/COL /		
SIP selection	PICS 5/3		
criteria:			
ISUP selection criteria:			
Test purpose:	Interworking of From header in the UPDATE. An additional connected number is sent in the ANM or CON		
	Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the "Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with		
	the option tag "from-change" is receive a Privacy header field was received an		set to PRIV_VALUE
	An UPDATE request is received containing with an identity in the format "+" CC+ NDC	+ SN then	· ·
	map the From header field received the ANM or CON	I in the UPDATE r	request to the Generic number in
	In the ANM or CON is included the Generi Number Qualifier = additional connecte		eter
	Address presentation restricted parame		on allowed
	Numbering plan indicator = ISDN number		on anowed
	Screening indicator = user provided, no		
	Address signals = derived from the Fro	m header in the L	IPDATE
SIP Parameter	INVITE: Supported: from-change		
values:	1XX or 2XX response: P-Asserted-Identity header field URI in the format "+"CC+NDC+SN,		
	Supported: from-change		
ISUP Parameter	UPDATE: From header in the format "+"CC+NDC+SN IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" =		
values:	requested"		
	ANM or CON		
	Genericnumber		
	Number Qualifier = "00000101"B Address presentation restricted parameter	_ '00'B	
	Nature of address indicator = '0000011'B	= 00 B	
	Numbering plan indicator = '001'B		
	Screening indicator = '11'B		
	Address signals = derived from the From h		
	ISUP	MGCF	SIP
	IAM →	→	
	CASE A	+	SIP_MESSAGE_VA
	CASE A		
	ACM ←	←	UPDATE
	ANM ←	→	_
	CASE B		
		+	*
	CON	→	200 OK UPDATE
		nversation	DVE
	REL →	→	
	RLC ←		200 OK BYE

TP606009	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clauses 7.4.2 and 7.5.2
TSS reference:	ISUP-SIP/SS/COL /	
SIP selection criteria:	PICS 5/22	

TP606009	SIP reference: RFC 320	61 [6]	ES 283 (ISUP reference: 027 [1], clauses 7.4.2 and 7.5.2
ISUP selection criteria:				
Test purpose:	Interworking of P-Asserted-Identity and From header. The Connected number and the additional connected number is presentation allowed			
	Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the "Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with			
	NDC+ SN and the option tag	the P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN and the option tag "from-change" is received a Privacy header field is not included		
	with an identity in the format "+"	CC+ NDC+	SN then	m header field containing a URI Frequest to the Generic number in
	In the ANM or CON is included Numbering plan indicator = I Address presentation restric Screening indicator = Netwo Address signals derived from	SDN/Telephoted parameter rk Provided	ony numbering er = Presentation	g plan ion allowed
	In the ANM or CON is included the Generic number parameter Number Qualifier = additional connected number Address presentation restricted parameter = Presentation allowed Numbering plan indicator = ISDN numbering plan Screening indicator = user provided, not verified Address signals derived from the From header in the UPDATE			
SIP Parameter values:	INVITE: Supported: from-change 1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN			
ISUP Parameter values:	UPDATE: From header, P-Asserted-Identity IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested"			
	ANM or CON Connected number parameter Address presentation restricted parameter = "00"B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT Generic number parameter Number Qualifier = "00000101"B Address presentation restricted parameter = "00"B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = derived from the From header in the UPDATE			
	ISUP		MGCF	SIP
	ISOP MIGCF SIP IAM → INVITE ← SIP_MESSAGE_VA			
	CASE A	_		
	ACM	+	•	← UPDATE
	ANM CASE B	←	7	→ 200 OK UPDATE
	CON	← Con		← UPDATE→ 200 OK UPDATE
	REL RLC	→ ←	-	→ BYE ← 200 OK BYE

TP606010	SIP reference: RFC 3261 [6]		ISUP reference: [1], clauses 7.4.2 and 7.5.2	
TCC reference:		L3 203 021	[1], Clauses 7.4.2 and 7.5.2	
TSS reference: SIP selection	ISUP-SIP/SS/COL / PICS 5/22			
criteria:	PICS 5/22			
ISUP selection criteria:				
Test purpose:	Interworking of P-Asserted-Identity to the connected number if no UPDATE was received			
	Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the "Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with			
	the P-Asserted-Identity header field cont NDC+ SN and the option tag "from-chan a Privacy header field is not included		an identity in the format "+" CC+	
	When the 200 OK was received, start timer	T _{TIR1}		
	An UPDATE request is not received			
	After T _{TIR1} was expired the ANM or CON is	sent		
	In the ANM or CON is included the Connected number Parameter .			
	Numbering plan indicator = ISDN/Telephony numbering plan			
	Address presentation restricted parameter = Presentation allowed Screening indicator = Network Provided			
	Address signals derived from the P-Asserted-Identity in the 200 OK INVITE			
SIP Parameter	INVITE: Supported: from-change	· · · · · ·		
values:	1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN			
ISUP Parameter	IAM: Optional Forward Call Indicators, Conr	ected Line Identity	y Request indicator" =	
values:	"requested"			
	ANIM			
	ANM or CON Connected number parameter			
	Address presentation restricted parameter =	: "00"B		
	Numbering plan indicator = '001'B			
	Screening indicator = '11'B			
	Address signals = PIXIT			
		MGCF	SIP	
	IAM →	→	INVITE SIP_MESSAGE_VA	
	CASE A			
	ACM ←	-		
	ANM ←	T _{TIR1}		
	CASE B			
		T _{TIR1}		
	CON	- 1 IIX I		
	1	nversation		
	REL →	→	BYE	
	RLC ←	+	200 OK BYE	

Values for tests purposes TP102005 to TP102010			
VA_01	VA_01 180 Ringing		
VA_02	VA_02 183 Session progress		
VA_03	200 OK		

	Values for test purpose TP102006 to TP102009			
VA PRIV_VALUE				
VA_1	ld			
VA_2	User			
VA_3	Header			

6.3.2.7 Sub-addressing (SUB)

TP607001	CW Ref	ference:	Selection criteria:
11 007 001	ES 283 027 [1],	clause 4.7.4.5.2	PICS 5/8
TSS reference:	ISUP-SIP/SS/SUB/		
Preconditions:			
Test purpose:	The calling party subaddress is mapped in the isub parameter of the P-Asserted-Identity		
	Ensure that the calling party subaddress in the ATP parameter of the received IAM is interworked in the isub parameter of the P-Asserted-Identity in the sent INVITE, if the Type of Subbaddress is set to "0 0 0" "NSAP.		
SIP Parameter	INVITE:		
values:	P-Asserted-Identit	y: sip: user part; isub=<	subaddress>@hostportion
ISUP Parameter values:	IAM: ATP(Calling party subaddress)		
Comments:	ISUP	MGCF	SIP
	IAM(ATP)	→	→ INVITE
	" " " " ' ' ' ' ' ' ' ' ' ' ' ' ' ' ' '	•	← 100 Trying
	ACM	←	← 180 Ringing
	ANM	←	← 200 OK INVITE
			→ ACK
		Commu	nication
	REL	→	→ BYE
	RLC	←	← 200 OK BYE

TP607002		eference: 7.4.5.2	Selection criteria: PICS 5/8	
TSS reference:	ISUP-SIP/SS/SUB/			
Preconditions:				
Test purpose:	The called party subaddress is mapped in the isub parameter of the Request URI			
	Ensure that the called party subaddress in the ATP parameter of the received IAM is interworked in the isub parameter of the Request URI in the sent INVITE, if the Type of Subbaddress is set to "0 0 0" "NSAP.			
SIP Parameter	INVITE:			
values:	Request URI: sip: user part; isub= <subaddress>@hostportion</subaddress>			
ISUP Parameter values:	IAM: ATP(Called party subaddress)			
Comments:	ISUP	MGCF	SIP	
	IAM(ATP)	→	→ INVITE	
			← 100 Trying	
	ACM	←	← 180 Ringing	
	ANM ← 200 OK INVITE			
			→ ACK	
		Commu	ınication	
	REL	→	→ BYE	
	RLC	<u>←</u>	← 200 OK BYE	

TD007000	CW Reference:		Selection criteria:	
TP607003	4.7.4.5.2		PICS 5/8	
TSS reference:	ISUP-SIP/SS/SUB/	•		
Preconditions:				
Test purpose:	The isub parameter of the P-Asserted-Identity in the 200 OK INVITE is mapped in the connected subaddress in the ANM			
	Ensure that the isub parameter in the P-Asserted-Identity of the received 200 OK INVITE is interworked in the Connected subaddress contained in an ATP parameter in the sent ANMOBCI. The Type of Subbaddress is set set to "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"			
SIP Parameter values:	INVITE: supported: from-change 200 OK INVITE: P-Asserted-IDENTITY: sip: user part; isub= <subaddress>@hostportion</subaddress>			
ISUP Parameter	IAM: oFCi: connected line request	•	•	
values:	ANM: ATP(Connected subaddress)			
Comments:	ISUP	MGCF	SIP	
	IAM →	-	NVITE	
		•	₹ 100 Trying	
	ACM ←	•	180 Ringing	
		-	÷ 200 OK INVITE • ACK	
	ANM(ATP) ←	•	UPDATE	
		-	▶ 200 OK UPDATE	
	Communication			
	REL →	-	▶ BYE	
	RLC ←	•	₹ 200 OK BYE	

TP607004		ference: .4.5.2	Selection criteria: NOT PICS 5/8	
TSS reference:	ISUP-SIP/SS/SUB/			
Preconditions:				
Test purpose:	The calling party subaddress is not mapped in the isub parameter of the P-Asserted- Identity Ensure that the calling party subaddress in the ATP parameter of the received IAM is not interworked in the isub parameter of the From header in the sent INVITE, if the Type of			
	Subbaddress is not se	et to "0 0 0" "NSAP.		
SIP Parameter	INVITE:			
values:	P-Asserted-Identity: sip: user part; isub= <subaddress>@hostportion</subaddress>			
ISUP Parameter values:	IAM: ATP(no Calling party subaddress)			
Comments:	ISUP MGCF SIP			
	IAM(ATP)	→	→ INVITE ← 100 Trying	
	ACM	←	← 180 Ringing	
	ANM ← 200 OK INVITE			
			→ ACK	
		Commu	inication	
	REL	→	→ BYE	
	RLC	-	← 200 OK BYE	

TP607005	_	Reference: .7.4.5.2	Selection criteria: NOT PICS 5/8			
TSS reference:	ISUP-SIP/SS/SUB/					
Preconditions:						
Test purpose:	The called party subaddress is not mapped in the isub parameter of the Request URI Ensure that the called party subaddress in the ATP parameter of the received IAM is not interworked in the isub parameter of the Request URI in the sent INVITE, if the Type of Subbaddress is not set to "0 0 0" "NSAP.					
SIP Parameter values:	INVITE: Request URI: si	p: user part; isub= <subac< td=""><td>ddress>@hostportion</td></subac<>	ddress>@hostportion			
ISUP Parameter values:	IAM: ATP(no Called	d party subaddress)				
Comments:	ISUP	MGCF	SIP			
	IAM(ATP)	→	→ INVITE			
			← 100 Trying			
	ACM	←	← 180 Ringing			
	ANM	←	← 200 OK INVITE			
		→ ACK				
		Comm	unication			
	REL	→	→ BYE			
	RLC	(← 200 OK BYE			

6.3.2.8 Closed user group (CUG)

TP608001	SIP refere	ence: RFC 3261 [6]		ISUP reference:
			ES	283 027 [1], clause 7.4.16
TSS reference:	ISUP-SIP/SS/CU	G/		
SIP selection	NOT PICS 5/7			
criteria:				
ISUP selection				
criteria:				
Test purpose:		s not support CUG, CUG	with outgoing	access allowed is interworked in a
	normal call.			
				orward call indicator, CUG call
				d CUG interlock code or CUG call
		•	nai iorward ca	Ill indicator is absent, the SIP
CID Domonoston		ure is not disrupted.		
SIP Parameter	No mapping			
values:				
ISUP Parameter values:				
	ICUD/DICC	CH.	-	SIP
Comments:	ISUP/BICC	, su		***
	IAM)	→	INVITE
	ACM	←	+	180 Ringing
		Ringing tone	_	
	ANM	+	←	200 OK INVITE
			→	ACK
		Cor	nversation	
	REL	→	→	BYE
	RLC	←	←	200 OK BYE

TP608002	SIP reference: R	FC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.4.16
TSS reference:	ISUP-SIP/SS/CUG/		20 200 027 [1], 010000 714110
SIP selection criteria:	NOT PICS 5/7		
ISUP selection criteria:			
Test purpose:	Ensure that the SUT if ar	n IAM is received w	with outgoing access not allowed is rejected. with Optional forward call indicator, CUG call coing access" and CUG interlock code, a REL cork.
SIP Parameter values:	No action		
ISUP Parameter values:	REL: Cause #29		
Comments:	ISUP/BICC IAM REL RLC	SUT → ← →	SIP

TP608003	SIP refere	nce: RFC 3261 [6]		ISUP reference: 7.4.1.2	
TSS reference:	ISUP-SIP/SS/CU	3/			
SIP selection	PICS 5/7				
criteria:					
ISUP selection criteria:					
Test purpose:	SIP network supp	orts CUG. CUG call indicate	or value "10"	received.	
				JG call indicator is mapped into	
				roup interlock code Parameter	
				tor> and the Binary code is	
		cug> <cuginterlockbinaryc< th=""><th>ode>.</th><th></th></cuginterlockbinaryc<>	ode>.		
SIP Parameter	INVITE:				
values:	<cug></cug>				
		ator>[derived from IAM Nety			
				ode]	
	<cugcommunicationindicator>10</cugcommunicationindicator>				
ISUP Parameter					
values:		Call Indicator CUG call indic	ootor - "10"		
values.	Closed User Grou		Saloi = 10		
		derived from INVITE XML bo	ndy - cualnt	erlockBinaryCode>	
		ntity derived from INVITE X			
Comments:	ISUP/BICC	SUT	in body in	SIP	
	IAM	→	→	INVITE	
	ACM	-	-	180 Ringing	
	1.5	Ringing tone	•		
	ANM	tunging tone	←	200 OK INVITE	
	, , , , , , ,	•	→	ACK	
		Conve	ersation	7.010	
	REL	→	→	BYE	
	RLC	←	ŕ	200 OK BYE	
	11120			ZOO ON DIL	

TP608004	SIP referer	nce: RFC 3261 [6]		ISUP reference: 7.4.1.2	
TSS reference:	ISUP-SIP/SS/CUG/				
SIP selection	PICS 5/7				
criteria:					
ISUP selection					
criteria:					
Test purpose:	SIP network suppo	orts CUG. CUG call indicato	r value "11"	received.	
	<pre><cug> < cugComm Network indentity</cug></pre>	nunicationIndicator>, the Clo	osed user gi tworkIndicat	JG call indicator is mapped into roup interlock code Parameter or> and the Binary code is	
SIP Parameter	INVITE:				
values:	<pre><cug> <networkindicator>[derived from IAM Network indentity]</networkindicator> <cuginterlockbinarycode>[derived from IAM Binary code]</cuginterlockbinarycode> <cugcommunicationindicator>11</cugcommunicationindicator> </cug></pre>				
ISUP Parameter	IAM:				
values:	Optional Forward (Call Indicator CUG call indic	ator = "11"		
	Closed User Group				
		erived from INVITE XML bo			
		tity derived from INVITE X	ML body < n		
Comments:	ISUP/BICC	SUT	_	SIP	
	IAM	→	→	INVITE	
	ACM	←	←	180 Ringing	
		Ringing tone			
	ANM	←	←	200 OK INVITE	
			→	ACK	
		Conve	rsation		
	REL	→	→	BYE	
	RLC	←	←	200 OK BYE	

6.3.2.9 Call diversion (CDIV)

TP609001	SIP refere	nce: RFC 3261 [6]		ISUP reference:
			ES	283 027 [1], clause 7.4.6
TSS reference:	ISUP-SIP/SS/ CD	IV /		
SIP selection	NOT PICS 5/12 A	ND NOT PICS 5/13 AND N	OT PICS 5/1	14 AND NOT PICS 5/15
criteria:				
ISUP selection				
criteria:				
Test purpose:	CDIV parameter r	not mapped		
				ng number, original called
	number and redi	rection information, contin	ue without c	lisrupting the SIP or ISUP
	signalling procedu	ıre.		
SIP Parameter	No mapping			
values:				
ISUP Parameter	IAM: Redired	cting number, Original called	I number, R	edirection information
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
		Ringing tone		
	ANM	*	←	200 OK INVITE
			→	ACK
		Conve	rsation	
	REL	→	→	BYE
	RLC		-	200 OK BYE
	NLC	<u> </u>		ZUU UI\ D I E

TP609002	SIP reference: RFC 3261 [6]	ISUP reference:			
		ES 283 027 [1], clause 7.5.4			
TSS reference:	ISUP-SIP/SS/ CDIV /				
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AT	ND PICS 5/15			
criteria:					
ISUP selection					
criteria:					
Test purpose:	IAM with Original call number and redirecting	number Presentation allowed received			
	Francisco that the CLIT if the LANA is received with	. Dedination number evision called			
	Ensure that the SUT if the IAM is received with				
	number Presentation allowed and redirection redirection counter value is "1", an INVITE is s				
	Redirecting number is contained in the in the l				
	number is contained in the hi-targeted-t				
	value in the latest history entry is mapped into				
SIP Parameter	INVITE: History-Info header	the realisation reason indicator.			
values:	hi-targeted-to-uri Redirecting number; index=1	or			
valuo01	hi-targeted-to-uri Redirecting number?Privacy				
	hi-targeted-to uri diverted to user; cause=Cau	se value: index=1.1			
ISUP Parameter	IAM:	- ,			
values:	Redirection information: "call diversion"				
	Redirection counter = 1				
	Redirecting indicator = 3				
	Redirecting reason = ISUP_RR				
	Original called number				
	Presentation restriction: Presentation allowed				
	Redirecting number				
Comments:	Presentation restriction: Presentation allowed	IT SIP			
Comments:	ISUP SU	_			
	IAM -	→ INVITE			
	ACM •	€ 180 Ringing			
	ANM ←	€ 200 OK INVITE			
	•	→ ACK			
	Commu				
	REL →	→ BYE			
	RLC	200 OK BYE			
		<u> </u>			

IAM		INVITE	
ISUP Parameter	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4			
TSS reference:	ISUP-SIP/SS/ CDIV /				
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15				
criteria:					
ISUP selection					
criteria:					
Test purpose:	IAM with Original call number Presentation reallwed received	stricted and redirecting number Presentation			
	Ensure that the SUT if the IAM is received with Redirecting number , original called number Presentation restricted and redirection information Presentation allowed, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1, the called party number is contained in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.				
SIP Parameter	INVITE: History-Info header				
values:	hi-targeted-to-uri Redirecting number?Privacy				
	hi-targeted-to uri diverted to user; cause=Cau	se_value; index=1.1			
ISUP Parameter	IAM:				
values:	Redirection information: "call diversion" Redirection counter = 1 Redirecting indicator = 3 Redirecting reason = ISUP_RR				
	Original called number Presentation restriction: Presentation restricte Redirecting number Presentation restriction: Presentation allowed				
Comments:	ISUP SU				
	IAM →	→ INVITE			
	ACM ←	← 180 Ringing			
	ANM ←	€ 200 OK INVITE			
		→ ACK			
	Commu				
	REL →	→ BYE			
	RLC ←	200 OK BYE ←			

IAM		INVITE	
ISUP Parameter or IE	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting		487
	'0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP609004	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4			
TSS reference:	ISUP-SIP/SS/ CDIV /	<u> </u>			
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14	AND PICS 5/15			
criteria:					
ISUP selection					
criteria:					
Test purpose:	IAM with Original call number Presentation allowed restricted	allowed and redirecting number Presentation			
	Ensure that the SUT if the IAM is received with Redirecting number , original called number Presentation allowed and redirection information Presentation restricted, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1, the called party number is contained in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.				
SIP Parameter	INVITE: History-Info header				
values:	hi-targeted-to-uri Redirecting number?Priva	cy=history; index=1,			
	hi-targeted-to uri diverted to user; cause=Ca	use_value; index=1.1			
ISUP Parameter	IAM:				
values:	Redirection information: "call diversion" Redirection counter = 1 Redirecting indicator = 4 Redirecting reason = ISUP_RR				
	Original called number				
	Presentation restriction: Presentation allowed	α			
	Redirecting number Presentation restriction: Presentation restriction:	ted			
Comments:		SUT SIP			
	IAM →	→ INVITE			
	ACM ←	← 180 Ringing			
	ANM ←	← 200 OK INVITE			
		→ ACK			
		unication			
	REL →	→ BYE			
	RLC	200 OK BYE			
	←	←			

IAM		INVITE	
ISUP Parameter or IE	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AI	ND PICS 5/15
criteria:		
ISUP selection		
criteria:		
Test purpose:	IAM with Original call number Presentation all allowed restricted	owed and redirecting number Presentation
	Ensure that the SUT if the IAM is received with number Presentation restricted and redirecting redirection counter value is "1", an INVITE is seedirecting number is contained in the in the launch number is contained in the intervalue in the latest history entry is mapped into	on information Presentation restricted, the sent containing a History-Info header. The ni-targeted-to-uri in index 1, the called party o-uri in index 1.1. The cause parameter
SIP Parameter	INVITE: History-Info header	
values:	hi-targeted-to-uri Redirecting number?Privacy hi-targeted-to uri diverted to user; cause= Cau	
ISUP Parameter	IAM:	
values:	Redirection information: "call diversion" Redirection counter = 1 Redirecting indicator = 4 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation restricte Redirecting number Presentation restriction: Presentation restricte	d
Comments:	ISUP SU	
	IAM →	→ INVITE
	ACM ←	← 180 Ringing
	ANM ←	← 200 OK INVITE
	_	→ ACK
	Commu	
	REL →	→ BYE
	RLC ←	200 OK BYE

IAM		INVITE	
ISUP Parameter or IE	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting		487
	'0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP609006	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /		
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 At	ND PICS 5/15	
criteria:			
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number and redirecting Redirecting indicator indicates "all redirection		
	Ensure that the SUT if the IAM is received with number Presentation allowed and redirection redirection counter value is "1", an INVITE is sedirecting number is contained in the in the linumber is contained in the hi-targeted-time.	n information Presentation restricted, the sent containing a History-Info header. The ni-targeted-to-uri in index 1, the called party pour in index 1.1. The cause parameter	
	value in the latest history entry is mapped into	the redirection reason indicator.	
SIP Parameter	INVITE: History-Info header		
values:	hi-targeted-to-uri Redirecting number; index=1		
	hi-targeted-to-uri Redirecting number?Privacy		
	hi-targeted-to uri diverted to user; cause=Cau	se_value; index=1.1	
ISUP Parameter	IAM:		
values:	Redirection information: "call diversion"		
	Redirection counter = 1		
	Redirecting indicator = 4 Redirecting reason = ISUP_RR		
	Redirecting reason = 150P_RR		
	Original called number		
	Presentation restriction: Presentation allowed		
	Redirecting number		
	Presentation restriction: Presentation allowed		
Comments:	ISUP SU	IT SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Commu	nication	
	REL →	→ BYE	
	RLC	200 OK BYE	
	+	+	

IAM		INVITE	
ISUP Parameter	Source value of parameter	SIP component	Derived value of
	field		header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting		487
	'0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP609007	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /	2,7	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AI	ND PICS 5/15	
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number and redirecting Redirection counter value 2	number Presentation allowed received,	
	Ensure that the SUT if the IAM is received with Redirecting number, original called number Presentation allowed and redirection information Presentation allowed, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the hi-targeted-to-uri in index 1.1, the called party number is contained in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the		
SIP Parameter	latest history entry is mapped into the redirect INVITE: History-Info header	ion reason indicator.	
values:	hi-targeted-to-uri Original called number; inde	x=1	
Turuoo:	hi-targeted-to-uri Redirecting number; cause=		
	hi-targeted-to-uri called party number; cause=		
ISUP Parameter	IAM:		
values:	Redirection information: "call diversion"		
	Redirection counter = 2		
	Redirecting indicator = 3		
	Redirecting reason = ISUP_RR		
	Original called number Presentation restriction: Presentation allowed		
	Presentation restriction: Presentation allowed		
	Redirecting number		
	Presentation restriction: Presentation allowed		
Comments:	ISUP SU		
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	ANM	€ 200 OK INVITE	
		→ ACK	
	Commu		
	REL →	→ BYF	
	RLC	200 OK BYE	
	←	200 OK BTE	
	<u> </u>		

IAM		INVITE	
ISUP Parameter or	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
ISUP_RR	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting		487
	'0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP609008	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /		
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15		
criteria:			
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number Presentation restricted and redirecting number Presentation allowed received, Redirection counter value 2		
	Ensure that the SUT if the IAM is received with Redirecting number , original called number Presentation restricted and redirection information Presentation allowed, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the hi-targeted-to-uri in index 1.1, the called party number is contained in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the		
SIP Parameter	latest history entry is mapped into the redirect INVITE: History-Info header	ion reason indicator.	
values:	hi-targeted-to-uri Original called number?Priva	acv-history; index-1	
values.	hi-targeted-to-uri Redirecting number; cause=		
	hi-targeted-to-uri called party number; cause=		
ISUP Parameter	IAM:		
values:	Redirection information: "call diversion"		
	Redirection counter = 2		
	Redirecting indicator = 3		
	Redirecting reason = ISUP_RR		
	Original called number	_	
	Presentation restriction: Presentation restricte	d .	
	Redirecting number		
	Presentation restriction: Presentation allowed		
Comments:	ISUP SU		
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	ANM ←	€ 200 OK INVITE	
		→ ACK	
	Commu		
	REL →	→ BYE	
	RLC	200 OK BYE	
	←	←	
	<u> </u>		

IAM		INVITE	
ISUP Parameter or	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
ISUP_RR	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609009	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /	1 1	
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15		
criteria:			
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number Presentation allowed and redirecting number Presentation restricted received, Redirection counter value 2		
	Ensure that the SUT if the IAM is received with Redirecting number , original called number Presentation allowed and redirection information Presentation restricted, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the hi-targeted-to-uri in index 1.1, the called party number is contained in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the		
SIP Parameter	latest history entry is mapped into the redirect INVITE: History-Info header	ion reason indicator.	
values:	hi-targeted-to-uri Original called number; inde	v–1	
values.	hi-targeted-to-uri Redirecting number; ?Privac		
	hi-targeted-to-uri called party number; cause=		
ISUP Parameter	IAM:	_ ,	
values:	Redirection information: "call diversion"		
	Redirection counter = 2		
	Redirecting indicator = 4		
	Redirecting reason = ISUP_RR		
	Original called number Presentation restriction: Presentation allowed		
	Fresentation restriction. Fresentation allowed		
	Redirecting number		
	Presentation restriction: Presentation restricte	d	
Comments:	ISUP SU		
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Commu		
	REL →	→ BYE	
	RLC	200 OK BYE	
	+	←	

IAM		INVITE	
ISUP Parameter or	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
ISUP_RR	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609010	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /	L3 203 021 [1], Clause 1.3.4	
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15		
criteria:	1100 0,127,110 1100 0,107,110 1100 0,117,11	100 0/10	
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number and redirecting number Presentation restricted received, Redirection counter value 2		
	Ensure that the SUT if the IAM is received with number Presentation restricted and redirection redirection counter value is "2", an INVITE is solved original called number is contained in the histonumber is contained in the in the history entry is mapped into the redirect	con information Presentation restricted, the sent containing a History-Info header. The atgeted-to uri in the index 1. The Redirecting co-uri in index 1.1, the called party number is ex 1.1.1. The cause parameter value in the	
SIP Parameter	INVITE: History-Info header	on reacon indicator.	
values:	hi-targeted-to-uri Original called number?Priva	acy=history index=1,	
	hi-targeted-to-uri Redirecting number; ?Privac		
	hi-targeted-to-uri called party number; cause=	Cause_value; index=1.1.1	
ISUP Parameter	IAM:		
values:	Redirection information: "call diversion"		
	Redirection counter = 2		
	Redirecting indicator = 4		
	Redirecting reason = ISUP_RR		
	Original called number		
	Original called number Presentation restriction: Presentation restricte	d	
	resentation restriction. Tresentation restricte		
	Redirecting number		
	Presentation restriction: Presentation restricte	d	
Comments:	ISUP SU	IT SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Commu		
	REL →	→ BYE	
	RLC	200 OK BYE	
	←	←	
	•		

IAM		INVITE	
ISUP Parameter or	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
ISUP_RR	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609011	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /		
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AI	ND PICS 5/15	
criteria:			
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number and redirecting number received, Redirection counter value 3		
	Ensure that the SUT if the IAM is received with Redirecting number , original called number and redirection information , the redirection counter value is "3", an INVITE is sent containing a History-Info header. The Original called number is contained in the hitatgeted-to uri in the index 1. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1.1, the called party number is contained in the in the hi-targeted-to-uri in		
	index 1.1.1. The cause parameter value in the	e latest history entry is mapped into the	
SIP Parameter	redirection reason indicator.		
values:	INVITE: History-Info header hi-targeted-to-uri Original called number; index=1,		
values.	hi-targeted-to-uri Dummy entry(PIXIT); cause:		
	hi-targeted-to-uri Redirecting number; cause= 486 ; index=1.1.1,		
	hi-targeted-to-uri called party number; cause=	Cause_value; index=1.1.1.1	
ISUP Parameter	IAM:		
values:	Redirection information: "call diversion"		
	Redirection counter = 3 Redirecting reason = ISUP_RR		
	Neuriecting reason = 130F_NN		
	Original called number		
•	Redirecting number	IT. OID	
Comments:	ISUP SU		
	IAM -	→ INVITE	
	ACM ← ANM ←	← 180 Ringing← 200 OK INVITE	
	AINIVI	→ ACK	
	Commu	nication	
	REL -	→ BYE	
	RLC	200 OK BYE	
	←	200 OK BTE	
		<u> </u>	

IAM		INVITE	
ISUP Parameter or	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
ISUP_RR	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609012	SIP reference: RFC 3261 [6]	ISUP reference:	
		ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /		
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15	
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number and redirecting number Presentation allowed received, Redirection counter value 2, Redirecting indicator indicates "all redirection information presentation restricted"		
	Ensure that the SUT if the IAM is received with Redirecting number , original called number Presentation allowed and redirection information Presentation restricted, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1.1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.		
SIP Parameter	INVITE: History-Info header		
values:	hi-targeted-to-uri Original called number?Privacy=history; index=1, hi-targeted-to-uri Redirecting number?Privacy=history; cause=302; index=1.1, hi-targeted-to-uri called party number; cause= Cause_value; index=1.1.1		
ISUP Parameter	IAM:	,	
values:	Redirection information: "call diversion" Redirection counter = 2 Redirecting indicator = 4 Redirecting reason = ISUP_RR Original called number		
	Presentation restriction: Presentation allowed Redirecting number		
	Presentation restriction: Presentation allowed		
Comments:	ISUP SU		
	IAM →	→ INVITE	
	ACM ←	 180 Ringing 	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Commu	nication	
	REL →	→ BYE	
	RLC	200 OK BYE	
	←	200 OK B1E	

IAM		INVITE	
ISUP Parameter or	Source value of parameter field	SIP component	Derived value of header field
Redirection Information	Redirecting reason	History-Info header	Cause parameter
ISUP_RR	unknown '0000'B	Cause_value	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609013	SIP reference: RFC 3261	[6]	ı	SUP reference:
			ES 283 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/CDIV/			
SIP selection				
criteria:				
ISUP selection	PICS 5/12 AND PICS 5/13 AND P	ICS 5/14 AI	ND PICS 5/15	
criteria:				
Test purpose:	181 Received, Notification subscription option according the Privacy header in the History- Info header			
	Ensure that the SUT (when no AC Forwarded) containing the History Info header, Privacy, priv-value co	-Index, Priva	acy, priv-value	component and the History-
	Sends an ACM message indicatin parameter coded	g a first dive	ersion with the E	Backward call indicators
	Called party's status indicator = no	indication		
	Redirection number:			
	Redirection number: derived from the Hi-target-to-uri of the last History-Info entry			
	the Call diversion information parameter			
	Notification subscription option = ISUP_NSO and the Generic notification indicator parameter = call is diverting			
SIP Parameter	181: History-Info header			
values:	hi-targeted-to-uri Redirecting number; index=1,			
	hi-targeted-to uri diverted to user; cause=Cause value?Privacy= priv-value ; index=1.1			
ISUP Parameter	ACM			
values:	BCI: No indication (00),			
	GenNot: Call is diverting (1111011	1),		
	Call diversion Info: ISUP_NSO			
	Redirection number: derived from			
Comments:	ISUP	SU		SIP
	IAM -		→	INVITE
	ACM ←		←	181 Being Forwarded
	CPG ←		←	180 Ringing
	ANM ←		-	200 OK INVITE
		•	→	ACK
	5	Commu		D)/E
	REL -		→	BYE
	RLC +			200 OK BYE

	SIP component History-Info header,	Call diversion information Notification subscription options
	priv-value component	ISUP_NSO
VA_01	Privacy header field absent	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "none"	ISUP_NSO = presentation allowed with redirection number
VA_03	Privacy "history"	ISUP_NSO = presentation allowed without redirection number

TP609014	SIP reference: RFC 3261 [6]	ISUP reference:	
TSS reference:	ICUD CID/CC/CDIV/	ES 283 027 [1], clause 7.5.4	
SIP selection	ISUP-SIP/SS/CDIV/		
criteria:			
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AI	ND DICC 5/15	
criteria:	PICS 5/12 AIND PICS 5/13 AIND PICS 5/14 AI	ND FICS 5/15	
Test purpose:	181 received ACM no indication is sent: NSC	Presentation not allowed	
rest purpose.	181 received, ACM no indication is sent: NSO Presentation not allowed		
	Ensure that the SUT (when no ACM has been sent before) on receipt of 181 (Call Is Being		
	Forwarded) containing priv-value history is se		
	redirecting uri and the diverted-to-uri then		
	Sends of an ACM message indicating a first d	iversion with the Backward call indicators	
	parameter coded		
	Called party's status indicator = no indication		
	Redirection number:		
	Redirection number: derived from the Hi-target-to-uri of the last History-Info entry		
	the Call diversion information parameter		
	Notification subscription option = presentation not allowed and the Generic notification indicator parameter = call is diverting		
SIP Parameter	181: History-Info header	neter = can is diverting	
values:	hi-targeted-to-uri Redirecting number? Privacy=history ; index=1,		
values.	hi-targeted-to-uri diverted to user; cause=Cause value? Privacy=history ; index=1.1		
ISUP Parameter	ACM		
values:	BCI: No indication (00),		
	GenNot: Call is diverting (1111011),		
	Call diversion Info: presentation not allowed		
	Redirection number: derived from the Hi-target-to-uri of the last History-Info entry		
Comments:	ISUP SU	JT SIP	
	IAM →	→ INVITE	
	ACM ←	 181 Being Forwarded 	
	CPG ←	← 180 Ringing	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	REL →	→ BYE	
	RLC ←	★ 200 OK BYE	

TP609015	SIP reference: RFC 3261 [6]		ISUP reference: 33 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/CDIV/	•	
SIP selection			
criteria:			
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/1	4 AND PICS 5/15	
criteria:			
Test purpose:	181 received sending of Redirection num	ber restriction para	meter in the ACM
	Ensure that the SUT (when no ACM has been sent before) on receipt of 181 (Call Is Being Forwarded) containing the History-Info header, Privacy, priv-value component Sends of an ACM message indicating a first diversion with the Backward call indicators parameter coded Called party's status indicator = no indication Redirection number: Redirection number: Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Redirection number restriction indicator Redirection number restriction option = ISUP_ReNrReIn and the Generic notification indicator parameter = call is diverting		
SIP Parameter	181: History-Info header		
values:	hi-targeted-to-uri Redirecting number; index=1,		
	hi-targeted-to uri diverted to user; cause=Cause value?Privacy= priv-value ; index=1.1		
ISUP Parameter	ACM:		
values:	BCI: No indication (00),		
	GenNot: Call is diverting (1111011),		
	Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Redirection number restriction indicator: ISUP_ReNrReIn		
Comments:	ISUP	SUT	SIP
Comments.	IAM →	301 →	INVITE
	ACM	-	
	CPG +	-	181 Being Forwarded
	ANM	-	180 Ringing 200 OK INVITE
	AINIVI	→	ACK
		7	ACN
	REL →	→	BYE
	RLC ←	←	200 OK BYE

	History-Info header Privacy, priv-value component	Redirection number restriction indicator ISUP_ReNrReIn
VA_01	Privacy "history"	Presentation restricted
VA_02	Privacy header field absent	Presentation allowed or absent
VA_03	Privacy "none"	Presentation allowed or absent

TP609016	SIP reference: RFC 3261 [6]	_	SUP reference: 3 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/CDIV/		
SIP selection criteria:			
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14	AND PICS 5/15	
Test purpose:	181 received, coding of notification subscription option in the CPG Progress		
	Ensure that the SUT, when an ACM has been sent before, on receipt of 181 (Call Is Being Forwarded) containing the History-Index, Privacy, priv-value component and the History-Info header, Privacy, priv-value component concerning the diverted-to uri Sends a CPG message indicating a first diversion with the Event information parameter coded		
	Event indicator = PROGRESS, Redirection number:		
	Redirection number: derived from the Hi-target-to-uri of the last History-Info entry the Call diversion information parameter Notification subscription option = ISUP_NSO and the Generic notification indicator parameter = call is diverting		
SIP Parameter values:	181: History-Info header hi-targeted-to-uri Redirecting number; index=1,		
ISUP Parameter	hi-targeted-to uri diverted to user; cause=Cause value?Privacy= Priv-value ; index=1.1 CPG:		
values:	Event indicator = PROGRESS, GenNot: Call is diverting (1111011), Call diversion Info: ISUP_NSO Redirection number: derived from the Hi-target-to-uri of the last History-Info entry		
Comments:	ISUP SUT SIP		
	IAM ACM CPG ANM ←	→ ← ← ←	INVITE 180 Ringing 181 Being Forwarded 200 OK INVITE ACK
	REL → RLC ←	→ ←	BYE 200 OK BYE

	SIP component History-Index Privacy, priv-value component	Call diversion information Notification subscription options ISUP_NSO
VA_01	Privacy header field absent	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "none"	ISUP_NSO = presentation allowed with redirection number
VA_03	Privacy "history"	ISUP_NSO = presentation not allowed,

TP609017	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/CDIV/	_ = =	
SIP selection			
criteria:			
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PI	ND PICS 5/15	
criteria:			
Test purpose:	181 received Privacy=history concerning the redirecting and the diverted-to URI setting of NSO in the CPG Progress		
	Ensure that the SUT, when an ACM has been Forwarded) containing priv-value history is serredirecting uri and the diverted-to-uri then		
	Sends a CPG message indicating a first divers	sion with the Event information parameter	
	Event indicator = PROGRESS,		
	Redirection number:		
	Redirection number: derived from the Hi-target-to-uri of the last History-Info entry		
	the Call diversion information parameter		
	Notification subscription option = presentation not allowed and the Generic notification indicator parameter coded		
	Notification indicator = call is diverting,		
SIP Parameter	181: History-Info header		
values:	hi-targeted-to-uri Redirecting number? Privacy=history ; index=1,		
	hi-targeted-to uri diverted to user; cause=Cause value? Privacy=history ; index=1.1		
ISUP Parameter	CPG:	,	
values:	Event indicator = PROGRESS,		
	GenNot: Call is diverting (1111011),		
	Call diversion Info: presentation not allowed		
Comments:	ISUP SU	_	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	CPG ←	 181 Being Forwarded 	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Commu		
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP092018	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/CDIV/	Lo 203 027 [1], clause 7.5.4	
SIP selection	1301 -311 /33/0017/		
criteria:			
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15		
criteria:	7 100 0, 127 11.12 1 100 0, 107 11.12 1 100 0, 10		
Test purpose:	181 received setting of Redirection number restriction in the CPG Progress		
	Ensure that the SUT, when an ACM has been sent before, on receipt of 181 (Call Is Being		
	der, Privacy, priv-value component diversion with the Event information		
	parameter coded Event indicator = PROGRESS, Redirection number: Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Redirection number restriction indicator= ISUP_ReNrReIn		
	and the Generic notification indicator parameter = call is diverting		
SIP Parameter	181: History-Info header		
values:	hi-targeted-to-uri Redirecting number; index=1,		
	hi-targeted-to uri diverted to user; cause=Ca	use value?Privacy= priv-value ; index=1.1	
ISUP Parameter	CPG:		
values:	Event indicator = PROGRESS,		
	GenNot: Call is diverting (1111011), Redirection number: ISUP_ReNr		
•	Redirection number restriction indicator: ISU		
Comments:		SIP SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	CPG ←	 181 Being Forwarded 	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Communication		
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

	History-Info header Privacy, priv-value component	Redirection number restriction indicator ISUP_ReNrReIn
VA_01	Privacy "history"	Presentation restricted
VA_02	Privacy "none"	Presentation allowed or absent
VA_03	Privacy header field absent	Presentation allowed or absent

TP609019	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4				
TSS reference:	ISUP-SIP/SS/CDIV/					
SIP selection criteria:						
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 A	ND PICS 5/15				
criteria:						
Test purpose:	180 received, CPG Alerting is sent, setting of NSO.					
	Ensure that the SUT on receipt of 180 (Ringing) (181 Call Is Being Forwarded was received before) containing the History-Info header, Privacy, priv-value component concerning diverted-to uri Sends a CPG message indicating a first diversion with the Event information parameter coded Event indicator = ALERTING, Redirection number: Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Notification subscription option = ISUP_NSO					
	and the Generic notification indicator parameter = call is diverting					
SIP Parameter	180: History-Info header					
values:	hi-targeted-to-uri Redirecting number; index=					
	hi-targeted-to uri diverted to user; cause=Cau	use value?Privacy= priv-value ; index=1.1				
ISUP Parameter	CPG:					
values:	Event indicator = ALERTING,					
	GenNot: Call is diverting (1111011), Redirection number: derived from the Hi-target-to-uri of the last History-Info entry					
	Notification subscription option: ISUP_NSO	et-to-un of the last History-into entry				
Comments:		UT SIP				
Comments.	IAM →	→ INVITE				
	ACM ←	← 181 Being Forwarded				
	CPG	← 180 Ringing				
	ANM ←	€ 200 OK INVITE				
	VIAINI	→ ACK				
	Commi	Inication				
	REL →	→ BYE				
	RLC ←	← 200 OK BYE				

	SIP component History-Info header, priv-value component	Call diversion information Notification subscription options ISUP_NSO
VA_01	Privacy header field absent	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "none"	ISUP_NSO = presentation allowed with redirection number
VA_03	Privacy "history"	ISUP_NSO = presentation allowed without redirection number

TP609020	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4				
TSS reference:	ISUP-SIP/SS/CDIV/					
SIP selection						
criteria:						
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15				
criteria:						
Test purpose:	180 received, CPG Alerting is sent, setting of NSO.					
	Ensure that the SUT on receipt of 180 (Ringing) (181 Call Is Being Forwarded was received before) containing the History-Info header, Privacy, priv-value component concerning redirecting and diverted-to uri Sends a CPG message indicating a first diversion with the Event information parameter coded Event indicator = ALERTING, Redirection number: Notification subscription option = presentation not allowed Redirection number: derived from the Hi-target-to-uri of the last History-Info entry and the Generic notification indicator parameter coded					
SIP Parameter	Notification indicator = call is diverting 180: History-Info header					
values:	hi-targeted-to-uri Redirecting number? Privacy	/=history: index=1.				
	hi-targeted-to uri diverted to user; cause=Cause					
ISUP Parameter	CPG:					
values:	Event indicator = ALERTING,					
	GenNot: Call is diverting (1111011),					
	Redirection number: derived from the Hi-targe					
_	Notification subscription option: Presentation r					
Comments:	ISUP SU	_				
	IAM →	→ INVITE				
	ACM ←	 181 Being Forwarded 				
	CPG ←	← 180 Ringing				
	ANM ←	← 200 OK INVITE				
		→ ACK				
	Commu	nication				
	REL →	→ BYE				
	RLC ←	← 200 OK BYE				

ICS 5/12 AND PICS 5/13 AND PICS BO received, CPG Alerting is sent, set Insure that the SUT on receipt of 18 Insure that th	tting of Redirection nu 30 (Ringing) (181 Call ry-Info header, Privacy	Is Being Forwarded was , priv-value component		
nsure that the SUT on receipt of 18 received before) containing the Historoncerning diverted-to uri Sends a CPG message indicating a parameter coded event indicator = ALERTING,	tting of Redirection nu 30 (Ringing) (181 Call ry-Info header, Privacy	Is Being Forwarded was , priv-value component		
nsure that the SUT on receipt of 18 received before) containing the Historoncerning diverted-to uri Sends a CPG message indicating a parameter coded event indicator = ALERTING,	tting of Redirection nu 30 (Ringing) (181 Call ry-Info header, Privacy	Is Being Forwarded was , priv-value component		
nsure that the SUT on receipt of 18 sceived before) containing the Historoncerning diverted-to uri Sends a CPG message indicating a sarameter coded event indicator = ALERTING,	30 (Ringing) (181 Call ry-Info header, Privacy	Is Being Forwarded was , priv-value component		
ceived before) containing the Histor oncerning diverted-to uri Sends a CPG message indicating a arameter coded vent indicator = ALERTING,	ry-Info header, Privacy	y, priv-value component		
edirection number restriction indicato nd the Generic notification indicator p otification indicator = call is diverting	-			
hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to-uri diverted to user; cause=Cause value?Privacy=priv-value; index=1.1				
CPG: Event indicator = ALERTING, Redirection number restriction indicator: ISUP_ReNrReIn				
EL →	→	SIP INVITE 181 Being Forwarded 180 Ringing 200 OK INVITE ACK BYE 200 OK BYE		
encis Fveildofn	direction number restriction indicator of the Generic notification indicator partification indicator partification indicator = call is diverting 0: History-Info header targeted-to-uri Redirecting number; targeted-to uri diverted to user; cause PG: ent indicator = ALERTING, endirection number restriction indicator UP M CM CM CM CO CL CD CD CD CD CD CD CD CD CD	edirection number: edirection number restriction indicator = ISUP_RNR d the Generic notification indicator parameter coded edification indicator = call is diverting 0: History-Info header targeted-to-uri Redirecting number; index=1, targeted-to uri diverted to user; cause=Cause value?Priva PG: ent indicator = ALERTING, edirection number restriction indicator: ISUP_ReNrReIn UP SUT M CM CG CC COMMUNICATION COMMUNICATI		

	History-Info header Privacy, priv-value component	Redirection number restriction indicator ISUP_ReNrReIn
VA_01	Privacy "history"	Presentation restricted
VA_02	Privacy "none"	Presentation allowed or absent
VA_03	Privacy header field absent	Presentation allowed or absent

TP609022	SIP reference: RFC 3261 [6]			ISUP reference: 3 027 [1], clause 7.5.4		
TSS reference:	ISUP-SIP/SS/CDIV/					
SIP selection criteria:						
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15					
Test purpose:	Redirection number restriction in ANM Ensure that the SUT on receipt of 200 (OK) containing the History-Info header, Privacy, priv-value component Sends a ANM message with Redirection number restriction indicator: ISUP_ReNrReIn					
SIP Parameter values:	200: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy=priv-value; index=1.1					
ISUP Parameter values:	ANM: Redirection number restriction indicate	or: ISUP_ReNrI	Reln			
Comments:	ISUP IAM ACM CPG ANM ←	SUT	+ + + + +	SIP INVITE 181 Being Forwarded 180 Ringing 200 OK INVITE ACK		
	REL → RLC ←		→	BYE 200 OK BYE		

	History-Info header Privacy, priv-value component	Redirection number restriction indicator ISUP_ReNrReIn
VA_01	Privacy "history"	Presentation restricted
VA_02	Privacy "none"	Presentation allowed or absent
VA_03	Privacy header field absent	Presentation allowed or absent

TP609023	SIP referen	nce: RFC 3261 [6]		ISUP reference: 3 027 [1], clause 7.5.4		
TSS reference:	ISUP-SIP/SS/CDI\	//				
SIP selection						
criteria:						
ISUP selection criteria:	PICS 5/12 AND PI	CS 5/13 AND PICS 5/14 A	AND PICS 5/15			
Test purpose:	Ensure that the SU	181 Received, no mapping to an ACM Ensure that the SUT (when no ACM has been sent before) on receipt of 181 (Call Is Being Forwarded) containing the History-Index no ACM is sent				
SIP Parameter		181: History-Info header				
values:	hi-targeted-to-uri R	tedirecting number; index=	₌ 1,			
	hi-targeted-to uri d	iverted to user; cause=Car	use value; index	x=1.1		
ISUP Parameter						
values:						
Comments:	ISUP	S	UT	SIP		
	IAM	→	→	INVITE		
			←	181 Being Forwarded		
	ACM	←	+	180 Ringing		
	ANM					
			→	ACK		
		Commi	unication			
	REL	→	→	BYE		
	RLC	+	←	200 OK BYE		

TP609024	SIP referer	nce: RFC 3261 [6]	1	ISUP reference: 3 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/CDIV	//			
SIP selection criteria:					
ISUP selection criteria:	PICS 5/12 AND PI	CS 5/13 AND PICS 5/14 AI	ND PICS 5/15		
Test purpose:	181 received, not mapped to a CPG Ensure that the SUT, when an ACM has been sent before, on receipt of 181 (Call Is Being Forwarded) containing the History-Index, no CPG is sent				
SIP Parameter	181: History-Info header				
values:	hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to-uri diverted to user; cause=Cause value?Privacy= Priv-value ; index=1.1				
ISUP Parameter values:	-				
Comments:	ISUP	SU		SIP	
	IAM	→	→	INVITE	
	ACM	←	+	180 Ringing 181 Being Forwarded	
	ANM	←	← →	200 OK INVITE ACK	
	REL	→	→	BYE	
	RLC		+	200 OK BYE	

TP609025	SIP reference: R	RFC 3261 [6]		ISUP reference: 3 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/CDIV/			
SIP selection criteria:				
ISUP selection criteria:	NOT PICS 5/12 AND NO	OT PICS 5/13 AND N	OT PICS 5/14	AND NOT PICS 5/15
Test purpose:	180 received, no mappin	ng.		
	Ensure that the SUT or received before) contain mapped			Is Being Forwarded was story-Info header is not
SIP Parameter	180: History-Info header			
values:	hi-targeted-to-uri Redired hi-targeted-to uri diverted			cy= priv-value ; index=1.1
ISUP Parameter values:				
Comments:	ISUP	SI	JT	SIP
	IAM	→	→	INVITE
			+	181 Being Forwarded
	ACM	←	←	180 Ringing
	ANM	←	+	200 OK INVITE
			→	ACK
		Commu	nication	
	REL	→	→	BYE
	RLC	←	+	200 OK BYE

TP609026	SIP reference	e: RFC 3261 [6]			ISUP reference: 3 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/CDIV/				
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 5/12 AND	NOT PICS 5/13 A	ND NOT PIC	S 5/14 /	AND NOT PICS 5/15
Test purpose:	No mapping of History-Info header in the 200 OK INVITE Ensure that the SUT on receipt of 200 (OK) containing the History-Info header, Privacy, priv-value component the History-Info header is not mapped				
SIP Parameter	200: History-Info header				
values:	hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to-uri diverted to user; cause=Cause value?Privacy=priv-value; index=1.1				
ISUP Parameter	No mapping				
values:					
Comments:	ISUP		SUT		SIP
	IAM	→		→	INVITE
				←	181 Being Forwarded
	ACM	←		←	180 Ringing
	ANM	←		←	200 OK INVITE
				→	ACK
	REL	→		→	BYE
	RLC			+	200 OK BYE

6.3.2.10 User to user signalling (UUS)

TP610001	SIP refere	ence: RFC 3261 [6]		ISUP reference:
			ITU-T Red	Q.1912.5 [32], annex B.21
				Q.737.1 [33], clause 1.1.7
TSS reference:	ISUP-SIP/SS/ UU	S/	1	
SIP selection				
criteria:				
ISUP selection criteria:				
Test purpose:	User-to-user service 1 implicit request not supported, User-to-user information discarded by the network			
	service 1 reques		cator in the	ser information as an implicit ACM "UUI discarded by the signalling procedure.
SIP Parameter	No mapping	· -		
values:				
ISUP Parameter		dicator "UUI discarded by the	e network", :	Service 1 response "No
values:	indication".			
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conve	ersation	
	REL	→	→	BYE
	RLC	←	←	200 OK BYE

TP610002	SIP refere	ence: RFC 3261 [6]			ISUP reference:	
			ITU	-T R	ec Q.1912.5 [32], annex B.21	
					Rec Q.737 [33], clause 1.1.7	
TSS reference:	ISUP-SIP/SS/ UU	S/				
SIP selection						
criteria:						
ISUP selection criteria:	PICS 11/1 AND P	PICS 11/2				
Test purpose:	User-to-user serv response	ice 1 explicit request	not essential i	not si	upported, service not provided	
	Ensure that the SUT if the IAM is received with an explicit service 1 request "Not essential" returns a User-to-user indicator in the ACM "Service 1 not provided" and continue without disrupting the SIP or ISUP signalling procedure.					
SIP Parameter	No mapping					
values:						
ISUP Parameter values:	ACM: User-to-ind provided"	dicator Service 1 resp	onse "Not			
Comments:	ISUP/BICC		SUT		SIP	
	IAM	→		→	INVITE	
	ACM	←		←	180 Ringing	
		Ringing tone				
	ANM	←		←	200 OK INVITE	
				→	ACK	
			Conversation	1		
	REL	→		→	BYE	
	RLC	←		←	200 OK BYE	

TP610003	SIP reference	e: RFC 3261 [6]	ISUP reference:
				ITU-T Rec Q.1912.5 [32], annex B.21
				ITU-T Rec Q.737.1 [33], clause 1.1.7
TSS reference:	ISUP-SIP/SS/ UUS /	1		
SIP selection				
criteria:				
ISUP selection criteria:	PICS 11/1 AND PIC	S 11/2		
Test purpose:	Ensure that the SUT	if the IAM is re	eceived with	itial not supported, rejected by sending a REL ith an explicit service 1 request "essential" stics containing the user-to-user indicator
SIP Parameter values:	No action			
ISUP Parameter	REL: cause #29, di	agnostics valu	e 0x2a	
values:				
Comments:	ISUP/BICC		SUT	SIP
	IAM	→		
	REL #29	←		
	RLC	→		

TP610004	SIP refere	ence: RFC 3261 [6]		ISUP reference:
				ec Q.1912.5 [32], annex B.21
			ITU-T	Rec Q.737 [33], clause 1.2.7
TSS reference:	ISUP-SIP/SS/ UL	JS /		
SIP selection				
criteria:				
ISUP selection criteria:	PICS 11/1 AND F	PICS 11/2		
Test purpose:	User-to-user serv response	rice 2 explicit request not	essential not s	upported, service not provided
	essential" returns		in the ACM "Se	it service 2 request "Not ervice 2 not provided" and ocedure.
SIP Parameter	No mapping			
values:				
ISUP Parameter	ACM: User-to-in	dicator Service 2 respons	e "Not provide	d"
values:	IOUD/DIOO	011	_	OID.
Comments:	ISUP/BICC	, su		SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
		Ringing tone	_	
	ANM	←	←	200 OK INVITE
		_	→	ACK
			nversation	
	REL	→	→	BYE
	RLC		<u> </u>	200 OK BYE

TP610005	SIP reference	e: RFC 3261	[6]	ISUP reference:
				ITU-T Rec Q.1912.5 [32], annex B.21
				ITU-T Rec Q.737.1 [33], clause 1.2.7
TSS reference:	ISUP-SIP/SS/ UUS /	1		
SIP selection				
criteria:				
ISUP selection	PICS 11/1 AND PICS	S 11/2		
criteria:				
Test purpose:	User-to-user service	2 explicit requ	uest essenti	tial not supported, rejected by sending a REL
	Ensure that the SUT	if the IAM is r	eceived with	ith an explicit service 2 request "essential"
	returns a REL with c	ause #29 and	an diagnos	stics containing the user-to-user indicator
	parameter name.			
SIP Parameter	No mapping			
values:				
ISUP Parameter	REL: cause #29, di	agnostics valu	ıe 0x2a	
values:				
Comments:	ISUP/BICC		SUT	SIP
	IAM	→		
	REL #29	(
	RLC	→		

TP610006	SIP refere	ence: RFC 3261 [6]			ISUP reference:	
			ITU	-T R	ec Q.1912.5 [32], annex B.21	
					ec Q.737.1 [33], clause 1.3.7.1	
TSS reference:	ISUP-SIP/SS/ UL	JS/				
SIP selection						
criteria:						
ISUP selection criteria:	PICS 11/1 AND F	PICS 11/2				
Test purpose:	User-to-user serv response	vice 3 explicit request	not essential	not si	upported, service not provided	
	Ensure that the SUT if the IAM is received with an explicit service 3 request "Not essential" returns a User-to-user indicator in the ACM "Service 3 not provided" and continue without disrupting the SIP or ISUP signalling procedure.					
SIP Parameter	No mapping					
values:						
ISUP Parameter values:	ACM: User-to-in	dicator, Service 3 res	ponse "Not pr	ovide	d"	
Comments:	ISUP/BICC		SUT		SIP	
	IAM	→		→	INVITE	
	ACM	←		←	180 Ringing	
		Ringing tone				
	ANM	←		←	200 OK INVITE	
				→	ACK	
			Conversation	1		
	REL	→		→	BYE	
	RLC	+		+	200 OK BYE	

TP610007	SIP reference:	RFC 3261 [6]		ISUP reference: T Rec Q.1912.5 [32], annex B.21 T Rec Q.737.1 [33], clause 1.3.7.1		
TSS reference:	ISUP-SIP/SS/ UUS /		•	<u> </u>		
SIP selection criteria:						
ISUP selection criteria:	PICS 11/1 AND PICS 1	1/2				
Test purpose:	User-to-user service 3 explicit request essential not supported, rejected by sending a REL Ensure that the SUT if the IAM is received with an explicit service 3 request "essential" returns a REL with cause #29 and an diagnostics containing the user-to-user indicator parameter name.					
SIP Parameter values:	No mapping					
ISUP Parameter values:	REL: cause #29, diagr	nostics value 0x2	a			
Comments:	ISUP/BICC IAM REL #29 RLC	S → ← →	UT	SIP		

TP610008	SIP referen	ce: RFC 3261 [6]		ISUP reference:
				Rec Q.1912.5 [32], annex B.21 Rec Q.737.1 [33], clause 1.3.7.1
TSS reference:	ISUP-SIP/SS/ UUS	/	1101	reco qui orri [co], ciaace ricirii
SIP selection criteria:				
ISUP selection criteria:	PICS 11/1 AND PIC	S 11/2		
Test purpose:	User-to-user service rejected by sending		not essential no	t supported in the confirmed state,
	Ensure that the SU- essential" returns a			olicit service 3 request "Not
SIP Parameter	No action			
values:				
ISUP Parameter	FRJ: User-to-user	indicator = "Service"	e 3 not provided	"
values:	101177700			
Comments:	ISUP/BICC	_	SUT	SIP
	IAM	→		→ INVITE
	ACM	←	•	← 180 Ringing
		Ringing tone		
	ANM	←	•	€ 200 OK INVITE
			•	→ ACK
			Conversation	
	FAR	→		
	FRJ	←		
			Conversation	
	REL	→		→ BYE
	RLC	←		€ 200 OK BYE

TP610009	SIP refere	nce: RFC 3261 [6]			ISUP reference:
11 01000			ITU	T R	ec Q.1912.5 [32], annex B.21
			'''		TU-T Rec Q.737.1 [33],
				•	clause 1.1.5.2.5.2.2
TSS reference:	ISUP-SIP/SS/ UU	S /			Oldudo IIIIo.z.io.z.iz
SIP selection	1001 011 7007 00	0 /			
criteria:					
ISUP selection	NOT PICS 11/2				
criteria:					
Test purpose:	User-to-user servi	ce 1 explicit reques	t not essential	not sı	upported, no response
	Encure that the SI	IT if the IAM is reco	vivad with an a	valici	t service 1 request "Not
					nalling procedure. No response
	to this request.	e without disrupting		ı sığı	naming procedure. No response
SIP Parameter	No mapping				
values:	. toappg				
ISUP Parameter					
values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	→		→	INVITE
	ACM	←		←	180 Ringing
		Ringing tone			
	ANM	←		←	200 OK INVITE
				→	ACK
			Conversation	1	
	REL	→		→	BYE
	RLC	←		+	200 OK BYE

TP610010	SIP refere	ence: RFC 3261 [6]			ISUP reference: ec Q.1912.5 [32], annex B.21 TU-T Rec Q.737.1 [33], clause 1.1.5.2.5.2.2
TSS reference:	ISUP-SIP/SS/ UU	IS /			Oldado IIIIoizioiziz
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 11/2				
Test purpose:	Ensure that the S		eived with an e x	xplici	it service 1 request "essential" cedure. No response to this
SIP Parameter	No action				
values:					
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	→		→	INVITE
	ACM	← Ringing tone		+	180 Ringing
	ANM	~		←	200 OK INVITE
				→	ACK
			Conversation	1	
	REL	→		→	BYE
	RLC	←		←	200 OK BYE

TP610011	SIP refere	ence: RFC 3261 [6]	ITU	-T R	ISUP reference: ec Q.1912.5 [32], annex B.21
					TU-T Rec Q.737.1 [33],
					clause 1.2.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UU	S/			
SIP selection					
criteria:					
ISUP selection	NOT PICS 11/2				
criteria:					
Test purpose:	User-to-user servi	ice 2 explicit reques	t not essential	not sı	upported, no response
					t service 2 request "Not
		e without disrupting	the SIP or ISU	P sig	nalling procedure. No response
OID Developed	to this request.				
SIP Parameter	No mapping				
values:					
ISUP Parameter values:					
	ISUP/BICC		CUT		SIP
Comments:			SUT		•
	IAM	→		→	INVITE
	ACM	←		←	180 Ringing
		Ringing tone			
	ANM	(←	200 OK INVITE
				→	ACK
			Conversation	1	
	REL	→		→	BYE
	RLC	←		←	200 OK BYE

TP610012	SIP refere	nce: RFC 3261 [6]			ISUP reference: ec Q.1912.5 [32], annex B.21 IU-T Rec Q.737.1 [33], clause 1.2.5.2.5.2.1	
TSS reference:	ISUP-SIP/SS/ UU	S /				
SIP selection criteria:						
ISUP selection	NOT PICS 11/2					
criteria:						
Test purpose:	User-to-user service 2 explicit request essential not supported, no response Ensure that the SUT if the IAM is received with an explicit service 2 request "essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.					
SIP Parameter	No action					
values:						
ISUP Parameter values:						
Comments:	ISUP/BICC		SUT		SIP	
	IAM	→		→	INVITE	
	ACM	←		←	180 Ringing	
		Ringing tone				
	ANM	+		←	200 OK INVITE	
				→	ACK	
			Conversation			
	REL	→		→	BYE	
	RLC	←		←	200 OK BYE	

TP610013	SIP refere	nce: RFC 3261 [6]			ISUP reference:
11 010010		100:111 0 0201 [0]	ITU	T R	ec Q.1912.5 [32], annex B.21
			110		TU-T Rec Q.737.1 [33],
				•	clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UUS	3 /			olduse 1.0.0.E.O.E.1
SIP selection	1001 011 /00/ 000	<i>31</i>			
criteria:					
ISUP selection	NOT PICS 11/2				
criteria:	110111001172				
Test purpose:	User-to-user servi	ce 3 explicit reques	t not essential	not su	upported, no response
	Ensure that the SI	IT if the IAM is rece	eived with an e x	nlici	t service 3 request "Not
					nalling procedure. No response
	to this request.	, manout diorupanig		o.g.	naming procedure. He respense
SIP Parameter	No mapping				
values:	3				
ISUP Parameter					
values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	→		→	INVITE
	ACM	←		←	180 Ringing
		Ringing tone			
	ANM	←		←	200 OK INVITE
				→	ACK
			Conversation		
	REL	→		→	BYE
	RLC	←		←	200 OK BYE

TP610014	SIP refere	nce: RFC 3261 [6]			ISUP reference: ec Q.1912.5 [32], annex B.21 IU-T Rec Q.737.1 [33], clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UU	S/			
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 11/2				
Test purpose:	Ensure that the S		eived with an ex	plici	t service 3 request "essential" cedure. No response to this
SIP Parameter	No action				
values:					
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	→		→	INVITE
	ACM	←		←	180 Ringing
		Ringing tone			
	ANM	←		←	200 OK INVITE
				→	ACK
			Conversation		
	REL	→		→	BYE
	RLC	+		+	200 OK BYE

TP610015	SIP refere	ence: RFC 3261 [6]		ISUP reference:
			ITU-T R	ec Q.1912.5 [32], annex B.21
			I ⁻	ГU-T Rec Q.737.1 [33],
				clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UL	IS /		
SIP selection				
criteria:				
ISUP selection	NOT PICS 11/1 C	OR NOT PICS 11/3		
criteria:				
Test purpose:	User-to-user serv	rice 3 explicit request not ess	ential not su	upported in the confirmed state,
	no response			
		UT if the FAR is received wit		
		e without disrupting the SIP	or ISUP sig	nalling procedure. No response
	to this request.			
SIP Parameter	No action			
values:				
ISUP Parameter				
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conve	rsation	
	FAR	→		
	REL	→	→	BYE
	RLC	-	-	200 OK BYE
ļ	1 2	_	<u> </u>	

TP610016	SIP refere	nce: RFC 3261 [6]			ISUP reference: ec Q.1912.5 [32], annex B.21 TU-T Rec Q.737.1 [33], clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UU	S/			
SIP selection					
criteria:					
ISUP selection criteria:					
Test purpose:	INVITE request Ensure that the SI	JT if the IAM conta	ins a User-to-us	er in	diser-to-User header field in the formation parameter, a User-to-idata component is derived from
SIP Parameter values:	INVITE: User-to-U	lser: uuidata derive	d from the User	-to-u	ser information
ISUP Parameter values:	IAM: User-to-user	information (PIXIT)		
Comments:	ISUP/BICC		SUT		SIP
	IAM	→		→	INVITE
	ACM	←		←	180 Ringing
		Ringing tone			
	ANM	←		←	200 OK INVITE
				→	ACK
			Conversation		
	REL	→		→	BYE
	RLC	←		←	200 OK BYE

TP610017	SIP refere	ence: RFC 3261 [6]		ISUP reference: ec Q.1912.5 [32], annex B.21 TU-T Rec Q.737.1 [33], clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UL	JS /		
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	parameter in the second that t	UT if the 180 Ringing conta neter is included in the ACM uuidata component. der starts with the first octe	ins a User-to	User-to-user information o-User header, a User-to-user User-to-user information is rotocol discriminator and followed
SIP Parameter values:	180: User-to-Use	r: uuidata derived from the l	Jser-to-user	information (PIXIT)
ISUP Parameter values:	ACM: User-to-use	er information		
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
		Ringing tone		
	ANM	←	←	200 OK INVITE
			→	ACK
		Conv	ersation	
	REL	→	→	BYE
	RLC		<u> </u>	200 OK BYE

TP610018	SIP referen	ice: RFC 3261 [6]		ISUP reference: ec Q.1912.5 [32], annex B.21 TU-T Rec Q.737.1 [33], clause 1.3.5.2.5.2.1	
TSS reference:	ISUP-SIP/SS/ UUS	5/			
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	User-to-user service 1 implicit response is mapped in the User-to-user information parameter in the ANM Ensure that the SUT if the 200 OK INVITE contains a User-to-User header, a User-to-user information parameter is included in the ANM and the User-to-user information is derived from the uuidata component. User-to-User header starts with the first octet being the protocol discriminator and followed by the user information octets				
SIP Parameter values:	200: User-to-User:	uuidata derived from the l	Jser-to-user	information (PIXIT)	
ISUP Parameter values:	ANM: User-to-user	information			
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→	→	INVITE	
	ACM	←	←	180 Ringing	
		Ringing tone			
	ANM	←	←	200 OK INVITE	
			→	ACK	
			ersation		
	REL	→	→	BYE	
	RLC	<u> </u>	<u> </u>	200 OK BYE	

TP610019	SID referer	nce: RFC 3261 [6]			ISUP reference:
11010019	SIF TELETE	ice. KFC 3201 [0]	17117	ГВа	ec Q.1912.5 [32], annex B.21
			110-1		U-T Rec Q.737.1 [33],
				"	clause 1.3.5.2.5.2.1
TSS reference:	IOUD OID/OO/ LILIG	<u> </u>			Clause 1.3.5.2.5.2.1
	ISUP-SIP/SS/ UUS	o /			
SIP selection					
criteria:	_				
ISUP selection criteria:					
Test purpose:	User-to-user servic	ce 1 implicit response i	s mapped in	the	User-to-user information
	parameter in the R		, ,		
		IT '(// D) /F / / ·			
					eader, a User-to-user information
			the User-to-t	user	information is derived from the
	uuidata component		atat balana ta		-4 -1:::
			ctet being the	e pro	otocol discriminator and followed
CID Devementer	by the user informa		41		information (DIVIT)
SIP Parameter values:	BYE: User-to-User	: uuidata derived from	the User-to-u	ıser	Information (PIXII)
	DEL : Usar ta ::sar	information			
ISUP Parameter values:	REL: User-to-user	information			
	ICUD/DIGO	0			CID
Comments:	ISUP/BICC	_	UT		SIP
	IAM	→		→	INVITE
	ACM	←		←	180 Ringing
		Ringing tone			
	ANM	←		←	200 OK INVITE
				→	ACK
		C	onversation		
	REL	←		←	BYE
	RLC	→		→	200 OK BYE

6.3.2.11 Explicit call transfer (ECT)

TP611001	SIP reference: RFC	3261 [6]	50	ISUP reference:	
			ES	283 027 [1], clause 7.4.8	
TSS reference:	ISUP-SIP/SS/ECT/				
SIP selection					
criteria:					
ISUP selection	PICS 12/1 AND NOT PICS	13/3			
criteria:					
Test purpose:	Loop prevention procedure supported, interworking of "call transfer" indication not supported				
	Ensure that the SUT if a LOP(request) is received returns a LOP (response) with the indication "insufficient information" continue without disrupting the SIP signalling procedure.				
	Ensure that the SUT if a FA procedure.	C is received cont	inue without	disrupting the SIP signalling	
SIP Parameter	No mapping				
values:					
ISUP Parameter	LOP: Response "insufficie	nt information"			
values:					
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→	→	INVITE	
	ACM	←	+	180 Ringing	
	Ring	ing tone			
	ANM	←		200 OK INVITE	
			→	ACK	
		Conve	ersation		
	LOP	→			
	LOP	←			
	FAC(Call transfer active)	→			
		Conve	ersation		
	REL	→	→	BYE	
	RLC	+	+	200 OK BYE	

TP611002	SIP reference: RFC	3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.4.8
TSS reference:	ISUP-SIP/SS/ECT/			
SIP selection criteria:				
ISUP selection criteria:	NOT PICS 12/1 AND NOT F	PICS 13/3		
Test purpose:	supported		· ·	"call transfer" indication not
	Ensure that the SUT if a LO signalling procedure. Ensure that the SUT if a FA procedure.	, ,		e without disrupting the SIP disrupting the SIP signalling
SIP Parameter values:	No mapping			
ISUP Parameter				
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	→	→	INVITE
	ACM	←	←	180 Ringing
	Ringi	ing tone		
	ANM	(←	200 OK INVITE
			→	ACK
		Conve	ersation	
	LOP	→		
	FAC(Call transfer active)	→		
		Conve	ersation	
	REL	→	→	BYE
	RLC		<u> </u>	200 OK BYE

TP611003	SIP reference: RFC	3261 [6]	ISUP reference: ES 283 027 [1], clause 7.4.8				
TSS reference:	ISUP-SIP/SS/ECT/						
SIP selection							
criteria:							
ISUP selection	PICS 12/1 AND PICS 13/3	PICS 12/1 AND PICS 13/3					
criteria:	1		1: (" " 500				
Test purpose:	Loop prevention procedure supported, interworking of "call transfer" indication in FAC supported						
	Ensure that the SUT if a LOP(request) is received returns a LOP (response) with the indication "insufficient information" continue without disrupting the SIP signalling procedure.						
	Ensure that the SUT if a FAC is received an INVITE is sent and the SDP contains an a-line set to "sendrecy".						
SIP Parameter	Re-INVITE SDP a=sendrecv	1					
values:							
ISUP Parameter	LOP: Response "insufficier	nt information"					
values:	FAC: Generic notification =	"call transfer, act	ive"				
Comments:	ISUP/BICC	SUT	SIP				
	IAM	→	→ INVITE				
	ACM	-	← 180 Ringing				
	Ringi	ng tone					
	ANM	←	← 200 OK INVITE → ACK				
	CPG(hold)	Conve	rsation → INVITE(sendonly) ← 200 OK INVITE(recvonly) → ACK				
	LOP LOP	→ ←	- / •				
	FAC(Call transfer, active)	→	→ INVITE(sendrecv)← 200 OK INVITE(sendrecv)→ ACK				
	551	Conve					
	REL	→	→ BYE				
	RLC		← 200 OK BYE				

TP611004	SIP reference: RFC	3261 [6]	ES 2	ISUP reference: 283 027 [1], clause 7.4.8	
TSS reference:	ISUP-SIP/SS/ECT/				
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 12/1 AND PICS	13/3			
Test purpose:	Loop prevention procedure supported, interworking of "call transfer" indication in FAC supported Ensure that the SUT if a FAC is received an INVITE is sent and the SDP contains an a-line				
SIP Parameter	set to "sendrecv". Re-INVITE SDP a=sendrec	<i>y</i>			
values:	The HAVITE OBT a Contained	Y			
ISUP Parameter values:	FAC: Generic notification =	= "call transfer, act	ive"		
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→	→	INVITE	
	ACM	←	←	180 Ringing	
	Ring	ing tone			
	ANM	←		200 OK INVITE ACK	
	CPG(hold)	Conve	←	INVITE(sendonly) 200 OK INVITE(recvonly) ACK	
	LOP FAC(Call transfer active)	→ →	→	INVITE(sendrecv)	
	REL RLC	Conve → ←	rsation → ←	BYE 200 OK BYE	

TP611005	SIP reference: RFC	3261 [6]	ES 2	ISUP reference: 283 027 [1], clause 7.4.8	
TSS reference:	ISUP-SIP/SS/ECT/				
SIP selection					
criteria:					
ISUP selection	PICS 13/3				
criteria:					
Test purpose:	Interworking of "call transfer	r" indication in CP0	G supported		
	Ensure that the SUT if a CPG Generic notification "call transfer, active" is received an INVITE is sent and the SDP contains an a-line set to "sendrecv".				
SIP Parameter values:	Re-INVITE SDP a=sendrecv				
ISUP Parameter	CPG: Generic notification =	= "call transfer, act	ive"		
values:					
Comments:	ISUP/BICC	SUT		SIP	
	IAM	→	→	INVITE	
	ACM	←	←	180 Ringing	
	Ring	ing tone			
	ANM	←	←	200 OK INVITE	
			→	ACK	
		Conve	rsation		
	CPG(hold)		→	INVITE(sendonly)	
			←		
			→	ACK	
	LOP	→			
	CPG(Call transfer active)	→	→	INVITE(sendrecv)	
			←	200 OK INVITE(sendrecv)	
			→	ACK	
		Conve	rsation		
	REL	→	→	BYE	
	RLC	+	<u> </u>	200 OK BYE	

6.3.2.12 Anonymous Call Rejection (ACR)

TP612001		B Reference: .7.1.3.1		Selection criteria:
TSS reference:	ISUP-SIP/SS/ACR			
Preconditions:				
Test purpose:	Mapping of 433 And	onymity Disallowed to REL	cause 24	
		B Anonymity Disallowed fina EL cause 24 " <i>call rejected</i> o		received to due the ACR service supplementary service"
SIP Parameter	433 Anonymity Disa	allowed		
values:				
ISUP Parameter values:	REL cause value 24	4 "call rejected due to ACR	' supplemer	ntary service"
Comments:	ISUP	MGCF		SIP
	IAM	→	→	INVITE
			←	100 Trying
	REL(24)	←	←	433 Anonymity Disallowed
	RLC` ´	→	→	ACK

TP612002	ACR-CB Refere 4.7.1.3.1	nce:	Selection criteria:		
TSS reference:	ISUP-SIP/SS/ACR				
Preconditions:					
Test purpose:	Mapping of 603 Decline to R	Mapping of 603 Decline to REL cause 21			
	Ensure that the 603 Decline final response received to due the ACR service is mapped into a REL cause 21 "call rejected"			the ACR service is mapped	
SIP Parameter values:	603 Decline				
ISUP Parameter values:	REL cause value 21 "call rejected"				
Comments:	ISUP	MGCF		SIP	
	IAM	→	→	INVITE	
			←	100 Trying	
	REL(24)	←	←	603 Decline	
	RLC	→	→	ACK	

6.3.2.13 Call waiting (CW)

FFS

6.3.2.14 Malicious call identification (MCID)

TP614001	MCID Reference:			Selection criteria:		
17014001	4.7.1.2			PICS 1/6		
TSS reference:	ISUP-SIP/SS/MCID/					
Preconditions:						
Test purpose:	Ensure that the XML mcid Mcid mapped into the MCID request	Mapping of XML mcid request (McidRequestIndicator) Ensure that the XML mcid McidRequestIndicator contained in a received INFO request mapped into the MCID request indicator requested in the sent IDR				
SIP Parameter values:	XML mcid request					
ISUP Parameter values:	IDR: MCID request indicator: M	IDR: MCID request indicator: MCID requested				
Comments:	ISUP	MGCF	•	SIP		
	IAM	→	→	INVITE 100 Trying		
	IDR(MCID request indicator)	←	←	INFO (XML mcid request) 200 OK INFO		
	ACM	←	←	180 Ringing		
	ANM	←	← →	200 OK INVITE ACK		
		Commun	ication	1		
	REL	→	→	BYE		
	RLC	←	←	200 OK BYE		

TP614002	MCID Reference: 4.7.1.2			Selection criteria: PICS 1/6 AND PICS 1/7		
TSS reference:	ISUP-SIP/SS/MCID/					
Preconditions:						
Test purpose:	Mapping of XML mcid request (HoldingIndicator) Ensure that the XML mcid HoldingIndicator is mapped into the MCID request indicator holding requested in the sent IDR					
SIP Parameter values:	INFO: XML mcid request HoldingIndicator = "1"					
ISUP Parameter values:	IDR: Holding indicator (national use): holding requested					
Comments:	ISUP		MGCF	SIP		
	IAM	→	→	INVITE		
			←	100 Trying		
	IDR(MCID request indicator)	←	←	INFO (XML mcid request)		
			→	200 OK INFO		
	ACM	←	+	180 Ringing		
	ANM	←	←	200 OK INVITE		
			→	ACK		
	Communication					
	REL	→	→	BYE		
	RLC	+		200 OK BYE		

TP614003	MCID Reference: 4.7.1.2			Selection criteria: PICS 1/6	
TSS reference:	ISUP-SIP/SS/MCID/				
Preconditions:					
Test purpose:	Mapping of IRS (McidResponseInc	dicator)			
	Ensure that MCID response indica		ntained i	n an IRS is mapped into the	
	XML mcid response McidRespons			• •	
	INFO:				
	XML mcid				
	request				
SIP Parameter	McidRequestIndicator =	: "1"			
values:	INFO:				
	XML mcid				
	response				
	McidResponseIndicator				
ISUP Parameter	IDR: MCID request indicator: MCII				
values:	IRS: MCID response indicator: MC				
Comments:	ISUP	MGCF		SIP	
	IAM	→	→	INVITE	
			←	100 Trying	
	IDR(MCID request indicator)	-	←	INFO (XML mcid request)	
			→	200 OK INFO	
	IRS (MCID response indicator)	→	→	INFO (XML mcid response)	
			←	200 OK INFO	
	ACM	←	←	180 Ringing	
	ANM	←	←	200 OK INVITE	
			→	ACK	
	Communication				
	REL	→	→	BYE	
	RLC	←	←	200 OK BYE	

TP614004	MCID Reference: 4.7.1.2			Selection criteria: PICS 1/6 AND NOT PICS 1/7		
TSS reference:	ISUP-SIP/SS/MCID/					
Preconditions:						
Test purpose:	Mapping of IRS (HoldingProvidedIndicator) Ensure that MCID response indicator holding provided, contained in an IRS is mapped into the XML mcid response HoldingProvidedIndicator.					
SIP Parameter	INFO: XML mcid request HoldingIndicator = "1"					
values:	INFO: XML mcid response HoldingProvidedIndicator = "0"					
ISUP Parameter	IDR: Holding indicator (national use)					
values:	IRS: Hold provided indicator (national					
Comments:	ISUP	MGCF		SIP		
	IDR(MCID request indicator)	→	→ ←	INVITE 100 Trying INFO (XML mcid request)		
	IRS (no MCID response indicator)		→ → ←	200 OK INFO INFO (XML mcid response) 200 OK INFO		
	ACM	←	←	180 Ringing		
	ANM	←	←	200 OK INVITE ACK		
	REL RLC	Communication → ←	on → ←	BYE 200 OK BYE		

TP614005	MCID Reference: 4.7.1.2			Selection criteria: PICS 1/6 AND PICS 1/7		
TSS reference:	ISUP-SIP/SS/MCID/					
Preconditions:						
Test purpose:	Mapping of IRS (HoldingProvidedIndicator) Ensure that MCID response indicator holding provided, contained in an IRS is mapped into					
	the YMI moid response HoldingProv	iidedladicator	(Holding	indicator is not for national		
	the XML mcid response HoldingProvidedIndicator (Holding indicator is not for national use).					
	INFO:					
	XML mcid					
	request					
SIP Parameter	HoldingIndicator = "1"					
values:	INFO:					
	XML mcid					
	response					
	HoldingProvidedIndicator = "1"					
ISUP Parameter	IDR: Holding indicator. holding requested					
values:	IRS: Hold provided indicator holding provided					
Comments:	ISUP	MGCI		SIP		
	IAM	→	→	INVITE		
		_	(100 Trying		
	IDR(MCID request indicator)	←	+	INFO (XML mcid request)		
			→	200 OK INFO		
	IRS (no MCID response indicator)		→	INFO (XML mcid response)		
			+	200 OK INFO		
	ACM	(←	180 Ringing		
	ANM	←	←	200 OK INVITE		
			→	ACK		
	Communication					
	REL	→	→	BYE		
	RLC	←	←	200 OK BYE		

Annex A (informative): Bibliography

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History

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